Splicecom maximiser

Installation & Reference Manual

Version 3.2
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Contents

Introduction

The **max**imiser business telephony system from SpliceCom provides a breakthrough in integrated voice communications. Developed from state of the art technology it delivers real life benefits associated with many traditionally separate components in one single, seamless system, currently supporting 4 to 10,000 IP or analogue extensions. When used in conjunction with SpliceCom's broad range of Proactive Communication Station (PCS) phones or applications, **max**imiser allows businesses to converge their telephone system with their core business applications, so ensuring that the right information can be delivered to the desktops of the right people at the right time.

Through the use of an innovative architecture, **max**imiser eliminates the physical and geographical limitations of traditional telephone systems, allowing great savings to be made on administration, management and infrastructure costs, through the unification of resource. This approach allows all businesses, irrespective of size, to benefit from extended communications, and more importantly protects their initial investment by growing with the business as the need for communication scales and becomes ever more demanding.

Embracing open standards wherever they exist, **max**imiser utilises a Linux operating system and IP communications as its core. This allows it to be deployed as a simple PBX with sophisticated, yet easy to use call handling facilities. Alternatively, **max**imiser can be used to deliver voice as just another application – albeit a very business critical one – in a fully converged IP network. In the latter case, **max**imiser's modular components can be distributed throughout a building, spread across a campus or even located in different locations, yet still operate and be managed as a single integrated system. With its ability to integrate seamlessly with multiple applications, independent of platform or location, **max**imiser delivers the blueprint for business telephony in the 21st Century.

The purpose of this manual is to give a step by step guide on how to install, configure and troubleshoot a **max**imiser system. The manual starts with a feature guide of each element of a system together with an understanding of the architecture used and finishes with a complete listing of each configuration field and its function together with in-depth technical data.

Please note that where the phrase "Call Server" is used this refers to the 5100 Call Server, the 5108 Call Server, the 4100 Call Server and the 4140 Remote Call Server unless specified.

Platform Overview

The **max**imiser system provides a single telephony system for small-to-medium sized companies with support for 4 to 10,000 extensions. It is a purely IP system supporting either traditional analogue or IP Phones.

5 Series Hardware

The 5 series **max**imiser range consists of the 5100 Call Server, 5300 Phone and 5108 Call Server.   Each module is 1u in height and can be mounted in a 19″ rack.

5100 Call Server

The main control unit of the system, this module provides:

- PBX
- Quad BRI (8 channels) / NT ports (2 ports enabled by licences)
- 2 x PRI (30 channels) / NT / DPNSS ports (1 channel on each port enabled, the remaining channels can be enabled by licences)
- 1 x LAN Link port, 10/100 Mbps Full Duplex Ethernet port for connectivity to existing LAN or dedicated LAN switch
- 4 x 10/100 Mbps Ethernet ports, integral QoS LAN switch, auto-sensing for MDI/MDIX connectivity and 802.3af Power over Ethernet supported
- 16 x standard analogue (POTS) ports with support for DTMF analogue handsets, PCS 510/505/10/5, fax machines and modems plus Caller Display support (8 ports enabled by licences)
- IP Router
- H.323 Gatekeeper
- H.323 Gateway
- Hard disk storage for software, LDAP database (supporting 10,000 configuration entries), voicemail, internal Web Server, Music on Hold and feature licenses
- 1 x power socket for 48 volts PSU supplied as standard
- 2 x Relay inputs for eg burglar alarm
- 2 x External relays for eg door entry system
- 1 x USB port for software upgrade purposes
- Supports H.450.2 (Call Transfer), H.450.4 (Hold), H.450.5 (Park) and H.450.7 (Message Waiting)
- SIP Proxy Server support
- Supports SIP and H.323 trunk services
- Supports NAT Traversal
- Native mode – G.711 64K with echo cancellation
- Can support up to 68 x G.729a Compression channels (requires licences)
- Supports 500 Users (analogue and/or IP)
- Up to 200 Call Servers per system (max 10,000 extensions, analogue and/or IP and max 5,000 trunks)

5300 Phone

This module provides connection for standard analogue telephones:

- Available as 5315 providing 15 POTS ports or 5330 providing 30 POTS ports
- Support for DTMF analogue handsets, PCS 5/10, fax machines and modems
- Caller Display support
- On-hook voltage 48V, off-hook current 25 mA, ring voltage 44V rms

- 4 x 10/100 Mbps Ethernet ports, integral QoS LAN switch, auto-sensing for MDI/MDIX connectivity and 802.3af Power over Ethernet supported
- 1 x LAN Link port, 10/100 Mbps Full Duplex Ethernet port for connectivity to existing LAN or dedicated LAN switch
- 1 x USB socket, allows wireless LAN connectivity via USB 2.0 WiFi adapter.
- 1 x power socket for 48 volts PSU supplied as standard
- Provided with PCS 60 or PCS 50 application to partner an analogue handset from a Windows or Mac OSX PC

5108 Call Server

The 5108 Call Server is designed either for the smaller business or the small remote office as part of a larger network. This module provides:
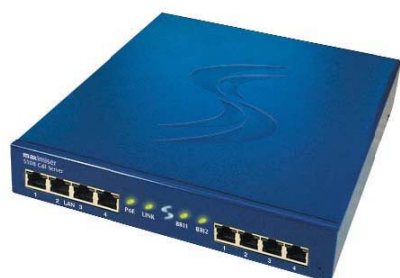
- PBX
- Dual BRI (4 channels) / NT ports (1 port enabled by licence)
- 4 x standard analogue (POTS) ports with support for DTMF analogue handsets, PCS 5/10, fax machines and modems plus Caller Display support
- IP Router
- 1 x LAN Link port, 10/100 Mbps Full Duplex Ethernet port for connectivity to existing LAN or dedicated LAN switch

- 4 x 10/100 Mbps Ethernet ports, integral QoS LAN switch, auto-sensing for MDI/MDIX connectivity and 802.3af Power over Ethernet supported
- H.323 Gatekeeper and H.323 Gateway
- Hard disk storage for software, LDAP database (supporting 10,000 configuration entries), voicemail, internal Web Server, Music on Hold and feature licenses
- 1 x Relay inputs for eg burglar alarm
- 1 x External relays for eg door entry system
- 1 x power socket for 48 volts PSU supplied as standard
- Supports H.450.2 (Call Transfer), H.450.4 (Hold), H.450.5 (Park) and H.450.7 (Message Waiting)
- SIP Proxy Server support
- Supports SIP and H.323 trunk services
- Supports NAT Traversal
- Native mode – G.711 64K with echo cancellation
- Can support up to 8 x G.729a Compression channels (requires licences)
- Supports 8 Users (or 12 Users with a 5108Plus licence – 4 analogue and 8 IP phone users, or 12 IP phone users)

4 Series Hardware

The 4 series **max**imiser system is made up of 3 components – 4100 Call Server, 4200 Trunk and 4300 Phone. The 4140 Remote Call Server combines the functionality of all three of these modules in a single unit. Each module is 1u in height and can be mounted in a 19" rack.

4100 Call Server

The main control unit of the system, this module provides:

- PBX
- Quad BRI (8 channels) / NT ports
- Optional single PRI (30 channels) / NT / DPNSS port
- IP Router
- 8 port QoS 10/100 Base-T switch
- Rear 10/100 LAN port
- WAN port V.11 - megastream 2 Mb

- H.323 Gatekeeper
- H.323 Gateway
- Hard disk storage for software (10 GB), LDAP database – supporting 10,000 User entries, voicemail, Internal Web Server, Music on Hold and feature licenses
- 48 volts PSU (so can purchase commercial UPS)
- 2 x Relay inputs for eg burglars alarms
- 2 x External relays
- Supports H.450.2 (Call Transfer), H.450.4 (Hold), H.450.5 (Park) and H.450.7 (Message Waiting)
- SIP Proxy Server support
- Supports SIP and H.323 trunk services
- Supports NAT Traversal
- LAN PWR port to provide Power over Ethernet (PoE) for the PCS 400/100.
- Native mode – G.711 64K with echo cancellation
- Optional Compression card supporting G.729 8K for WAN links
- Supports 300 Users (analogue and/or IP)
- Supports 4 Trunk modules
- Up to 100 Call Servers and Trunk modules per system (max 5,000 extensions, analogue and/or IP)

4200 Trunk

This module provides additional ISDN and WAN ports:

- Quad BRI (8 channels) / NT ports
- Optional single PRI (30 channels) / NT / DPNSS port
- WAN port V.11 - megastream 2 MB
- 8 port QoS 10/100 Base-T switch
- Integral H.323 Gateway
- IP Router
- LAN Power port for Power over Ethernet

- Up to 4 Trunk Modules per Call Server
- Up to 100 Trunk Module and Call Servers per system

4300 Phone

This module provides connection for standard analogue telephones:

- Available as 4315 providing 15 POTS ports or 4330 providing 30 POTS ports
- Support for DTMF analogue handsets, PCS 5/10, fax machines and modems
- Caller Display support
- On-hook voltage 48V, off-hook current 25 mA, ring voltage 44V rms

- 10/100 LAN port for connection to network
- PCMCIA slot for wireless LAN connection
- Provided with PCS 60 or PCS 50 application to partner an analogue handset from a Windows or Mac OSX PC

4140 Remote Call Server

The 4140 Remote Call Server is designed either for the smaller business or the small remote office as part of a larger network.

- PBX
- 2 x BRI (4 channels) / NT ports
- Optional additional 2 x BRI (4 channels)/NT ports
- Optional PRI (15 channels) / NT / DPNSS port
- 8 x POTS ports
- IP Router
- 1 port QoS 10/100 Base-T switch
- IP WAN port V.11 - megastream 2 Mb (for connection to a Call Server only)
- H.323 Gatekeeper
- H.323 Gateway
- Hard disk storage for software (10 GB), LDAP database, voicemail, Internal Web Server and feature licenses
- 48 volts PSU (so can purchase commercial UPS)
- Supports H.450.2 (Call Transfer), H.450.4 (Hold), H.450.5 (Park) and H.450.7 (Message Waiting)
- SIP Proxy Server support
- Supports SIP and H.323 trunk services
- Supports NAT Traversal
- Native mode – G.711 64K with echo cancellation
- Optional Compression card supporting G.729 8K for WAN links
- Supports 40 Users (8 analogue plus a further 32 analogue and/or IP)

4400 Voice Compression Module

As an alternative to the 64kbps, G.711 voice supported as standard on maximiser, 8kbps, G.729a based encoding, provides an excellent balance between voice quality and bandwidth efficiency. The 4400 Voice Compression Module provides 16 channels, of 8 kbps, G.729a based compression and can be deployed to provide a higher density of calls between 4100 Call Servers/4140 Remote Call Servers over low speed WAN links.

- 1 x RJ45 port, dual speed, 10/100 Mbps FDX Ethernet interface with integral LEDs for Link and Data.
- 1 x 48Vdc Power jack.
- Supports G.729a
- Additional 16 channels enabled via a licence, giving a maximum of 32 compression channels per module

maximiser Handsets

PCS 580G

This IP phone provides:

- Standard telephony functionality
- Graphical colour touch screen LCD interface
- Context Sensitive Screen
- Ability to display real time video and graphical information during call
- Power - 802.3af Power over Ethernet
- 2 Port 10/100/1000 Mbps FDX LAN switch – supporting automatic MDI-MDIX crossover, Diffserv Quality of Service, 802.1q VLAN and 802.3af Power over Ethernet
- 2 x USB port – USB 2.0 interface for WiFi and external keyboard
- Headset connection
- Wide-angle tilting keypad/display panel
- Message waiting/do not disturb "S" LED
- Dual Mode – SIP (configured via the Call Server) & H.323
- Fully integrated with the LDAP system database for viewing and annotating customer records and notes
- Integral web browser
- Can be wall mounted
- Can be partnered with PCS 60 or PCS 50

PCS 570

This IP phone provides:

- Standard 12 button telephony functionality
- Full colour, backlit, graphics display (240 x 320) with auto-dimming, context sensitive
- Wide-angle tilting keypad/display panel
- 18 intuitive, multi-functional context sensitive keys
- 10 fixed function keys
- Power - 802.3af Power over Ethernet
- 2 Port 10/100 Mbps FDX LAN switch – supporting automatic MDI-MDIX crossover, Diffserv Quality of Service, 802.1q VLAN and 802.3af Power over Ethernet
- USB port – USB 2.0 interface for WiFi and external keyboard
- Headset connector
- Message waiting/do not disturb "S" LED
- Dual Mode – SIP (configured via the Call Server) & H.323
- Fully integrated with the LDAP system database for access to the directories of Users, Departments and Contacts.
- Can be wall mounted
- Can be partnered with PCS 60 or PCS 50

PCS 570G

This IP phone provides the same functionality as the PCS 570 except it has a Gigabit interface providing a 2 Port 10/100/1000 Mbps FDX LAN switch.

PCS 560

This IP phone provides:

- Standard 12 button telephony functionality
- Full colour, backlit, graphics display (240 x 320) with auto-dimming, context sensitive
- Wide-angle tilting keypad/display panel
- 9 intuitive, multi-functional context sensitive keys
- 10 fixed function keys
- Power - 802.3af Power over Ethernet
- 2 Port 10/100 Mbps FDX LAN switch – supporting automatic MDI-MDIX crossover, Diffserv Quality of Service, 802.1q VLAN and 802.3af Power over Ethernet
- Headset connector
- Message waiting/do not disturb "S" LED
- Dual Mode – SIP (configured via the Call Server) & H.323
- Fully integrated with the LDAP system database for access to the directories of Users, Departments and Contacts.
- Can be wall mounted
- Can be partnered with PCS 60 or PCS 50

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PCS 410

This IP Phone provides:

- Standard telephony functionality
- Graphical touch screen LCD interface
- Context Sensitive Screen
- Ability to display real time video and graphical information during call
- Power over Ethernet support – can be powered via a LAN port on the Call Server, via the network, or via the optional dedicated PSU
- 2 Port 10/100 Mbps FDX LAN switch
- USB port for connection of mouse and keyboard
- Internal USB for WiFi
- Headset connection
- Fully integrated with the LDAP system database for viewing and annotating customer records and notes
- Integral web browser
- Can be wall mounted
- Can be partnered with PCS 60 or PCS 50

PCS 400

This IP Phone provides:

- Standard telephony functionality
- Graphical touch screen LCD interface
- Ability to display real time video and graphical information during call
- Context Sensitive Screen
- Power over Ethernet support – can be powered via the network, via the Powered Ethernet PSU or optional dedicated PSU
- Provides LAN port for PC
- 802.11b PCMCIA Card Slot for Wireless LAN
- USB port for connection of mouse and keyboard
- Headset connection
- Fully integrated with LDAP system database for viewing and annotating customer records and notes
- Integral web browser
- Can be wall mounted
- Can be partnered with PCS 60 or PCS 50

PCS 100

The PCS 100 is an IP Phone presented as a conventional business telephone. It provides caller ID, a dedicated Busy Lamp Field together with pre programmed buttons for simple and easy access to system features.

- Standard 12 button telephony functionality
- Display of number/name
- 64 x 128 Context Sensitive LCD Display
- 8 Dynamic, Context Sensitive Keys
- Headset connection.
- Powered via the LAN or optional STEPS power supply
- Fully integrated with system database for viewing and speed dialling users, departments, contacts and favourites
- 2 Port QOS switch, single cable to desk.
- Can be wall mounted
- Enhanced volume and speech quality of incoming calls with Automatic Gain Control (AGC).
- Can be partnered with PCS 60 or PCS 50

PCS 520

The PCS 520 is an analogue handset providing the following features:

- Standard 12 button telephone keypad
- 3 line LCD display with contrast settings
- Display of Calling Line Number/Name & Called Line Name
- Dual message waiting/DND indicator
- Visual ringing indicator
- 10 pre-programmed keys
- 10 user definable keys
- Internal directory (100 names/numbers)
- Missed call/Clear/Dial/Pause/Recall/Redial/Speaker/Volume /Mute keys
- Voicemail guide on handset
- Desktop paging
- Auto Answer
- Headset port
- Powered via the line port
- Can be partnered with PCS 60 or PCS 50

PCS 10

The PCS 10 is an analogue handset providing the following features:

- Standard 12 button telephone keypad
- 3 line LCD display with contrast settings
- Display of Calling Line Number/Name & Called Line Name
- Message Waiting icon
- 10 pre-programmed keys
- 10 user definable keys
- Internal directory (100 names/numbers)
- Missed call/Clear/Dial/Pause/Recall/Redial/ Speaker/Volume/Mute keys
- Headset port
- Powered via a dedicated PSU or with batteries
- Can be partnered with PCS 60 or PCS 50

PCS 505

The PCS 505 is an analogue handset providing the following features:

- Standard 12 button keypad
- Visual call indicator/Message Waiting Lamp
- Headset socket
- Hold/Recall key
- Last number Redial key
- Microphone Mute key
- Headset key
- System Feature Guide
- Voicemail management keys
- Can be partnered with the PCS 60 or PCS 50

PCS 5

The PCS 5 is an analogue handset providing the following features

- Standard 12 button keypad
- Visual call indicator/Message Waiting Lamp
- Recall/Redial/Mute/Pause keys
- System Feature Guide
- Voicemail management keys
- Can be partnered with the PCS 60 or PCS 50

Power over Ethernet PSU

This is an external power supply for use with the 4100 Call Server and 4200 Trunk Module:

- Providing IP phones, including the PCS 400/100, with power over the LAN when connected to the integral 10/100 Mbps switch.

- It is connected to LAN PWR port on the back of the 4100 Call Server and/or 4200 Trunk Module
- The LAN PWR port provides power to 2 spare pairs on Ethernet ports
- Up to 8 IP Phones can be powered via the Powered Ethernet PSU.
- Supports the 802.3af Power over Ethernet standard

PCS PSU

This is an external power supply providing the PCS 400 IP phones with local power for applications not using power over LAN, eg Wireless LAN connected phones.

Single Terminal Ethernet Power Supply (STEPS)

This is an External Power over Ethernet PSU for the PCS 580, 570, 560, 410, 400 and100 or other 802.3af compliant devices.

Please note:
- if a PCMCIA card is required in the PCS 400 a standard PCS PSU must be used.
- If used with a PCS 580G or PCS 570G the LAN port (port 1) will not run at 1G.

Voice Compression Card

maximiser supports G.711 encoding with echo suppression as standard for transporting voice. Where compressed voice is a requirement, G.729a between Call Servers is supported through optional internal Voice Compression Cards, one fitted at each end. This feature will support up to 8 channels.

Please refer to page 326 for further technical data on these products.

maximiser Software

PCS 60

This PC application provides the follow facilities:
- Standard telephony functionality
- CLI display
- Context sensitive toolbar
- Fully integrated with the system database for speeding dialling, viewing and annotating customer records and notes
- Ability to display real time video and graphical information during a call
- Operator Console Mode available with an Operator Console licence
- Can be run as either an IP soft phone or as a partner to a PCS 580, 570, 560, 410/400, 100, 10, 5 or an alternative analogue handset.
- Available for MS Windows and Apple Mac OS X v10.4 or above

Please refer to the PCS 60 section from page 58 for further details.

PCS 50

This PC application provides the same telephony functionality as available on the PCS 410/400. PCS 50 can run as an IP Soft phone or be programmed to partner a User's PCS 5/10/100/560/570/580 or existing analogue phone from their PC.

Versions for Windows and Mac OS X are currently available.

Please refer to the PCS 50 section from page 69 for further details.

PCS Operators Console

This application combines all of the main PCS 50 functions on a single full screen page for operators and receptionist. Like the PCS 50 from which it is derived it can be run as an IP soft phone or as a Partner to PCS 580, 570, 560, 410/400, PCS 100, PCS 10, PCS 5 or existing analogue phones.

Windows and Mac OS X versions are available.

Please refer to the PCS Operator Console section from page 71 for further details.
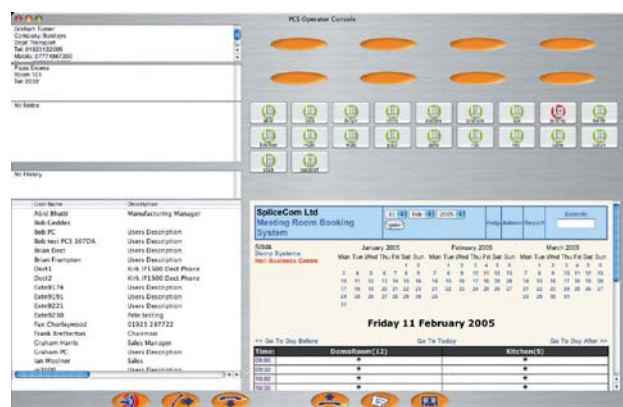
Manager

This is a web-based application that is used to view and configure the database. It can be run from any web browser (IE v5 or higher, Netscape v6 or higher, Apple Safari) that can connect to the **max**imiser.

Please refer to the Manager section from page 24 for further details.
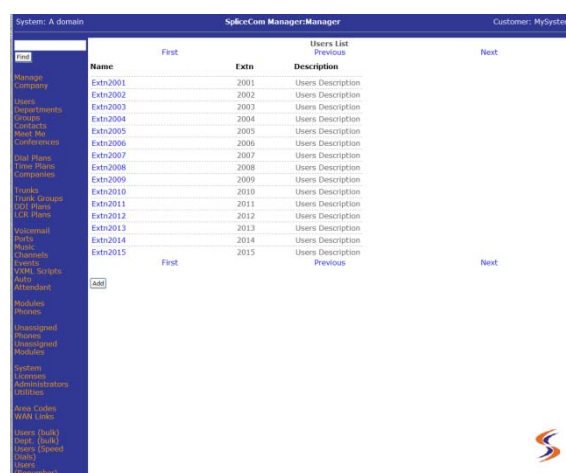
Voicemail

The 5100, 5108, 4100 Call Servers and 4140 Remote Call Server all provide integral voicemail & Auto-Attendant facilities.

- A 5100 Call Server provides up to 16 simultaneous connections, 4 of which are licensed on shipment, and 1500 hours of message storage
- A 5108 Call Server provides up to 8 simultaneous connections, 2 of which are licensed on shipment, and 500 hours of message storage
- A 4100 Call Server provides up to 8 simultaneous connections and 30 hours of message storage
- A 4140 Remote Call Server provides up to 4 simultaneous connections and 10 hours of message storage

This voicemail and Auto-Attendant application can also be run on a standalone Linux PC or Mac PC where more simultaneous connections or greater storage capacity is required.

A MessageBox licence is required for each User and Department requiring a voicemail facility.
A Voice Processing Port licence is required for each additional simultaneous connection.

Please refer to the Working with Voicemail section from page 187 for further details.

Enhanced Speech Processing (ESP)

Enhanced Speech Processing is SpliceCom's advanced multi-layer auto-attendant and Interactive Voice Response (IVR) system for **max**imiser. Licensed on a per-port basis ESP utilises standards based VoiceXML to deliver great flexibility and interoperability for voice processing requirements. ESP allows you to answer calls, play files, collect DTMF digits, record files and transfer calls. Features and functionality supported by ESP include;

- Multi-Layer Auto Attendant
- Dial-by Extension
- Voice Forms (DTMF digit and spoken response capture)
- Text-To-Speech (TTS) Conversion
- Reminder/Wake-Up/Alarm Calls
- IMAP Unified Messaging
- Spoken Email Collection & Reply
- Queue Buster
- Call Back When Free

For further information please refer to the Introduction to ESP document available within the Partner area of the SpliceCom website.

Vision

SpliceCom Vision is a web based application suite developed from the ground up by SpliceCom to work with **max**imiser – and to only work with **max**imiser. Utilising the latest AJAX and Web 2.0 technologies, Vision has been designed to deliver business critical information, in an easy to understand manner, wherever and whenever it's needed. Vision offers four services; Reports (historical), Recording (capture) , Live (real-time) & Mobility (freedom).

For full details on how employ this application suite with your **max**imiser system please refer to the Vision Installation and Configuration Manual.
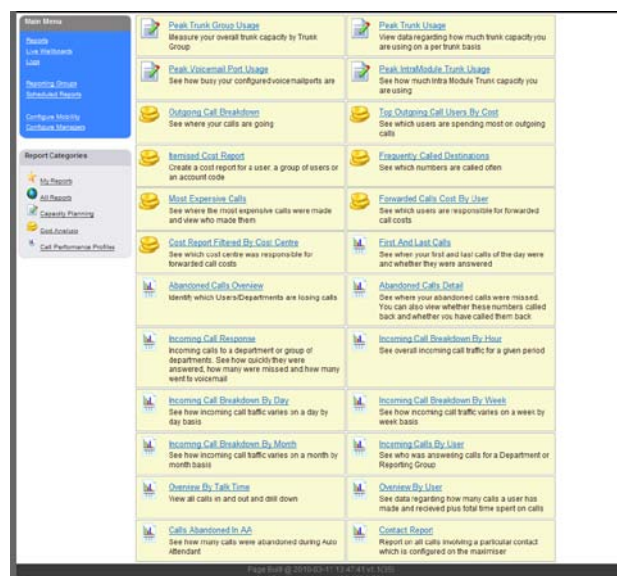
System Architecture

The system operates entirely around a LAN/IP network environment, eliminating any previous geographic constraints imposed upon similar systems. The system is constructed with several distinct and discrete components, all physically independent and operating symbiotically to provide a single seamless system.

Working with the System

All **max**imiser modules are linked via TCP/IP. Each module is a host on the network with its own IP address. Therefore each module can be connected via standard CAT 5 cabling, leased line, or Wireless LAN networking in the case of the Phone Module and the PCS 400. This allows for multi-site networking with the advantage of only one system to configure & manage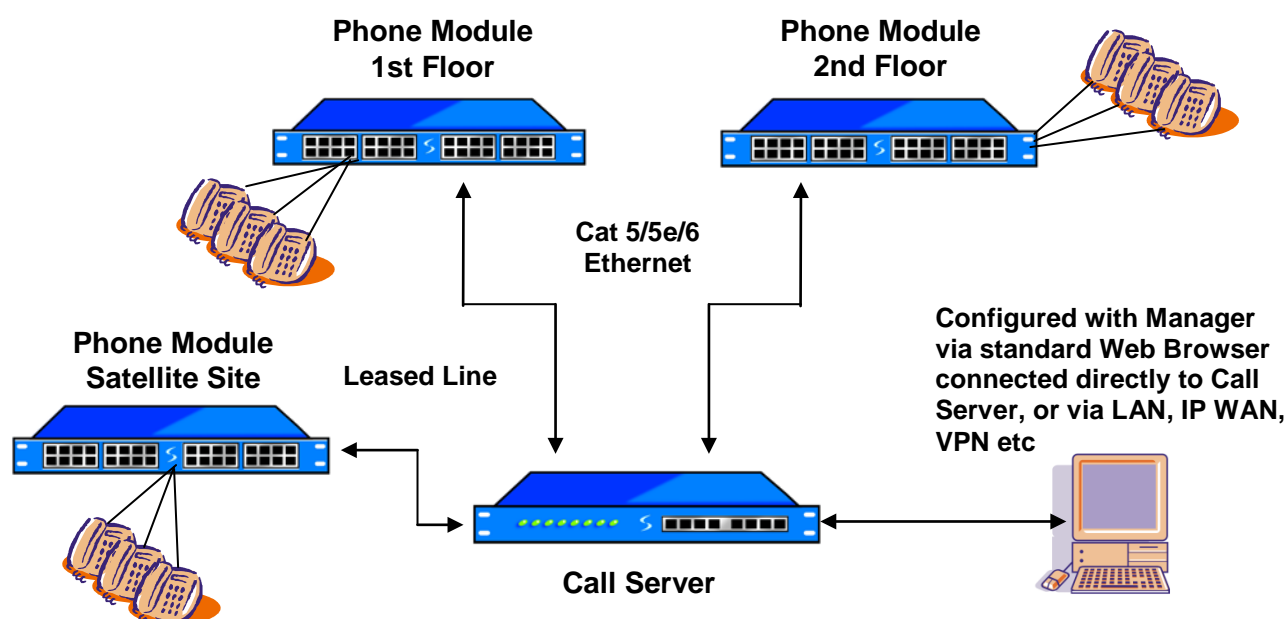 – from anywhere. The configuration of the system is stored on the database which resides on each Call Server; this can be accessed via Manager. Manager is a web-based application so can be run from any web browser that has an IP connection to the system.

As the **max**imiser system is a purely IP system all calls are carried over IP therefore H.323, the industry standard for Voice over IP, is supported. This means that the Call Server acts as the H.323 Gateway. A gateway is responsible for translating one transmission type into another, so in the case of the Call Server, for example, it will translate calls received via ISDN so they can travel over an IP network and vice a versa. The Trunk Module will also provide this functionality. The Call Server also acts as the H.323 Gatekeeper and is responsible for controlling the access into and out of the network – phone registration, call routing and call logging etc. Quality of Service, also managed by the Call Server, will ensure that Voice packets have priority over Data packets. This will make sure that a conversation is not broken up or interrupted.

The configuration of the system is stored on the LDAP database held on the hard disk contained within the Call Server. LDAP (Lightweight Directory Access Protocol) is the protocol used to store information on a database that can be searched and queried over a network by any number of users. In the case of the **max**imiser system LDAP allows the configuration to be viewed and amended via Manager and allows users to view telephone directories and contact information via a PCS 580, 570/560, PCS 410/410, PCS 100, PCS 60 and PCS 50. As LDAP is the industry standard for accessing databases future applications will include access to third party databases.

**Phone Module
1st Floor**

**Phone Module
2nd Floor**

**Cat 5/5e/6
Ethernet**

**Configured with Manager
via standard Web Browser
connected directly to Call
Server, or via LAN, IP WAN,
VPN etc**

**Phone Module
Satellite Site**

Leased Line

Call Server

System Directory

The system is powered by a centrally replicated but distributed information directory. In essence this allows components of the system that are on different continents to act and operate in the same way as components only inches away from the central unit.

All systems access this database using LDAP (the internet standard for accessing databases) for both reading and writing to the information.

Call Servers are responsible for holding, distributing and replicating this information. All other modules must have a direct connection via a LAN to a Call Server.

There are two different scenarios that are constructed:

Primary Mode
The Call Server that acts as the Primary administration module and holds the master database for the entire system. This Call Server is responsible for replicating this database to other Secondary Call Servers.

Secondary Mode
Secondary Mode indicates that this Call Server will receive a replica of the database from the Primary Call Server. Any local changes are passed back to the Primary for distribution to all other Secondary Call Servers.

In all of the above cases non-administrative modules, ie Phone and Trunk modules, obtain their configuration from a local Call Server, thus they do not need to be configured directly. This allows live deployment of new modules with minimum complexity.

Gatekeeper

All Trunks and Phones register with the Gatekeeper in the usual H.323 secure methods. The Gatekeeper is responsible for all Phone Registrations, Call Routing and Call Logging.

Although communication peers (eg phones) pass the voice data directly between them, the Gatekeeper will control all of the call routing. The Gatekeeper therefore does not require high performance data capabilities.

The Call Server runs Gatekeeper functionality.

Trunks

Trunk gateways are registered with the Gatekeeper and act as a resource for the Gatekeeper to route calls to and from the public PSTN networks. Trunks can be on physically separate modules and geographic sites.

Clock synchronisation is performed by a combination of buffering and "Clock Count" signalling between the master system and all slaves.

Call Servers and Trunk Modules run Gateway functionality.

Distributed Phone Modules

As each Phone Module is LAN connected it can now be distributed geographically within a building or separate buildings and will only require one cable between the module(s) and the administration system. This is particularly useful when wiring different floors of a building; only a small number of vertical risers are needed. Each 100 Mpbs FDX Ethernet connection is capable of handling between 390-780 concurrent calls, assuming no other traffic. Where non-QoS (Quality of Service) Switches are deployed then it is recommended that Voice be separated from normal LAN traffic, each Call Server and Trunk module contains a QoS Switch to allow such segregation. In normal circumstances (simple telephony) it is recommended that the Ethernet cabling be treated as a new generation of flexible voice cabling and independently connected, especially with the use of structured wiring and patch panels.

Computer Telephony

Because each Phone Module is geographically independent so is the Phone oriented computer telephony. Each Phone Module is responsible for its own handling of a third party TAPI interface.

Additionally the central administrator can have first party telephony ports via TAPI.3.0 and H.323. This is the method used for external voicemail.

Distributed Voicemail

Because voicemail utilises H.323, voicemail platforms can be operating system independent and located in any appropriate location. Because of the multi-port registration philosophy voicemail can be spread across many machines increasing the potential capacity of the system. This is especially important when SAPI (speech recognition) is required.

Call Servers are capable of running an Auto attendant and voicemail service as standard. Voicemail can also run on a Linux server and will handle storage limited only by the disk space of the machine (approx 0.5 Mbytes per minute).

Extendibility

Although the **max**imiser platform provides primary applications; Proactive Communication Station, voicemail etc, the use of H.323 and LDAP means that independent vendors can hook into the system extending the range of options available. This is especially important in the Call Centre market.

Manageable Objects

The internal LDAP database contains the following configurable objects.

Users
Each user can make use of the system for communications. These are the primary parameters for all of the telephony facilities that a user may have. They are distinct from the phone parameters since a user may login at any phone. Please refer to the Working with Users section from page 80 for further details.

Departments
Departments are used to determine the distribution of calls to a particular group of Users. A caller can dial one number and any available member of a Group can answer the call. A Department is used to determine what will happen to the call out of hours, if the members of the Group are unavailable or busy. Please refer to the Routing calls via a Department section from page 128 for further details.

Groups
Groups are collections of users and/or capabilities that can be used to identify groups of users. A distribution group can contain just one user, a thousand users or no users. Alternatively it may have a capability requirement to automatically select users with a particular skill eg French 80%. Groups may contain other groups to form "super groups". Please refer to the Working with Groups section from page 122 for further details.

Contacts
The Contacts database provides directory lookup information of, eg customers and suppliers. An incoming call will be alpha tagged if a CLI match is found in the Contacts database and will be displayed on the PCS, or caller display of the phone. Contacts may also be marked with account codes for billing and tracking purposes. Please refer to the Setting up the Contacts Database section from page 141 for further details.

Meet Me Conferences
A Meet Me Conference provides the facility to create a conference that can be joined by Users as and when required. Please refer to the Conferencing section from page 151 for further details.

Dial Plans
Dial Plans determine how digits dialled on a User's handset are handled. Please refer to the Working with Dial Plans section from page 114 for further details.

Time Plans
Time Plans provide a positive restriction of when activities may happen. In essence a Time Plan defines when things are allowed to operate by day of the week and time of the day. Please refer to the Using a Time Plan section from page 148 for further details.

Companies

Users, Departments, Contacts and Meet Me Conferences can be assigned to different Companies so that in a multi-tenancy system Users are unaware of the usage of the system by other businesses. An Auto Attendant can be assigned to each Company. Please refer to the Working with Companies section from page 155 for further details.

Trunks

Trunks are the method used for outgoing or incoming communication with the system whether via ISDN, SIP, or H.323. Each Trunk port available on the system can be configured individually. Please refer to the Working with Trunks section from page 159 for further details.

Trunk Groups

A Trunk group allows the grouping together of multiple Trunks for delivery of calls. This is very useful in geographically organising the flow of calls, and can also be used to differentiate levels of service, by keeping spare capacity for high priority calls. Please refer to the Working with Trunks section from page 161 for further details.

DDI Call Plans

Direct Dial In Call Plans determine how incoming ISDN calls will be routed. Please refer to the Working with DDI Plans section from page 169 for further details.

LCR Plans

LCR Plans apply to all calls that will be routed to a trunk via a gateway. It allows the best selection of a trunk and selection of service providers, based upon the number dialled and time of day. It can also be used to control the geographic exit point from the network, eg all calls to London numbers would exit via a London trunk module, but calls to Edinburgh would exit via the Edinburgh trunk module. If a trunk should be busy or fail then alternate translations and trunks would be used. Please refer to the Working with LCR Plans section from page 175 for further details.

Voicemail Ports

A Voicemail Port determines the facilities provided by a Call Server's voicemail service. Where more than one Call Server is active on a system Voicemail Ports allow different capabilities to be assigned to each Call Server providing a voicemail service.   Please refer to the Working with Voicemail section from page 188 for further details

Music Channels

Music Channels determine how music on hold is played when external calls are placed on hold either from an internally stored WAV file or via an external source. Please refer to the Configuring Music Channels section from page 177 for further details.

Events

Events allow web pages to be sent to a PCS 60, PCS 50 or PCS 410/400 to display warning messages, company information etc. A Relay Input port or a Dial Plan entry can activate an Event. Please refer to the Using Events section from page 181 for further details.

VXML Scripts

VXML Scripts provide the ability to create multi-level auto-attendants incorporating features such as availability based routing, dial by extension, out of hours routing and text-to-speech. Please refer to the Creating VXML Scripts section from page 225 for further details.

Auto Attendant

Voicemail provides the ability to create a simple Auto Attendant facility whereby callers can be transferred to a User or Department from a predefined list of options. Please refer to the Creating an Auto Attendant section from page 222 for further details.

Modules

Modules determine the components that make up the **max**imiser system. Each module can be configured individually. Please refer to the Connecting a Module section from page 39 for further details.

Phones

Phones are the analogue ports and IP Phones connected to the system. Each phone can be configured individually. Please refer to the Connecting an Analogue Extension section from page 77 and the Connecting a PCS section from page 51 for further information.

Unassigned Phones

An Unassigned Phones is an IP Phone requesting registration to the system. Please refer to the Using Auto Add Phones section from page 34 for further details.

Unassigned Modules

An Unassigned Module is a module requesting registration to the system. Please refer to the Connecting a Module section from page 39 for further details.

System

This object gives access to system functionality and system setting such a System Name required for licence activation and the system Locale. Please refer to the Configuring System Details section from page 26 for further details.

Licences

Licences allow the use of a specific facility on the system or increase the specification of the system. Each licence purchased is exclusive to one system and one function. Please refer to the Installing Licences section from page 35 for further details.

Administrators

The list of administrators of the system. A default Administrator called "Manager" will be created at start-up with read/write access to all objects, and will be given a password of "managerpassword". Please refer to the Setting Up Administrative Access section from page 31 for further details.

Utilities

Facilities available for managing, maintaining and troubleshooting the system. Please refer to the Utilities section from page 250 for further details.

Area Codes

Ambiguous telephone numbers to be matched with incoming CLI when a match is not found within the Contacts database. The main purpose of these entries is to match the area code part of the CLI so that caller display will inform the User from which area the call has originated. Please refer to the Using Area Codes section from page 177 for further details.

WAN Links

WAN Links provide the ability to connect remote systems via a 2Mbps Megastream leased line or via ISDN dial up. Please refer to the Data Routing section from page 247 for further details.
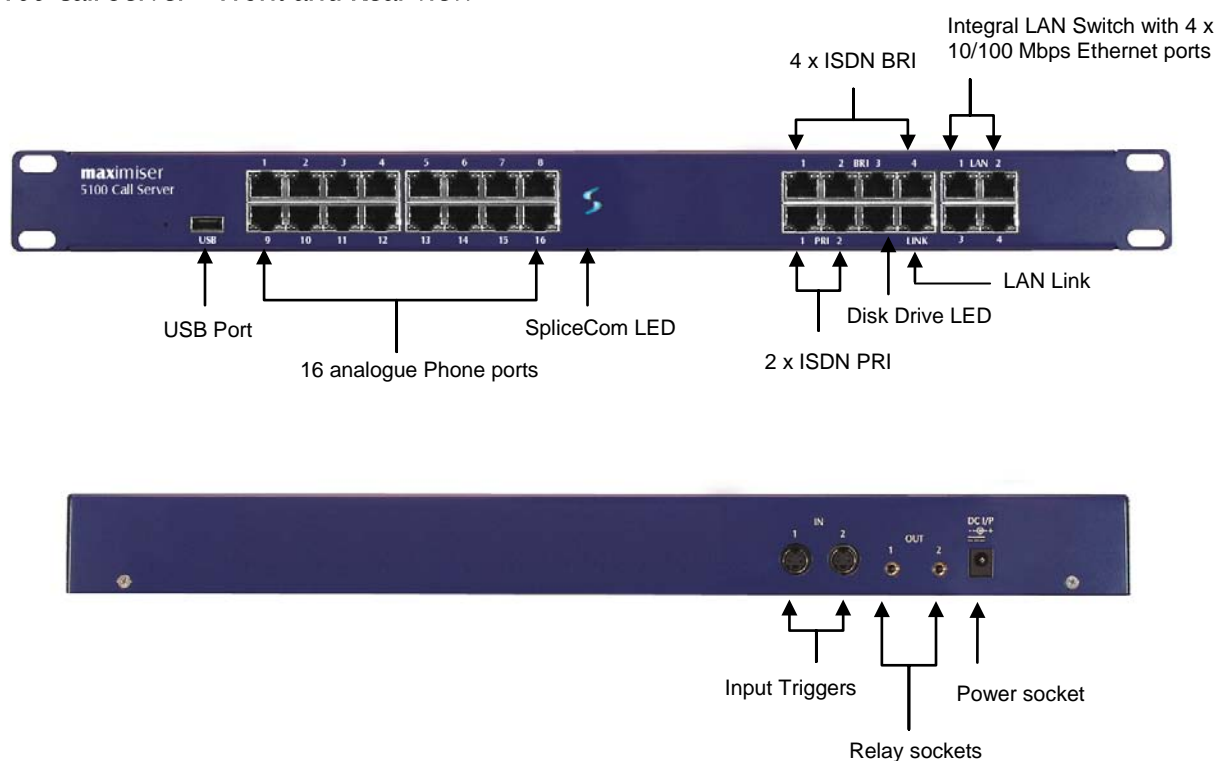
Installing a System

A 5100 or 5108 Call Server or 4100 Call Server or 4140 Remote Call Server should be the first module to be powered up; all subsequent modules will join this system.
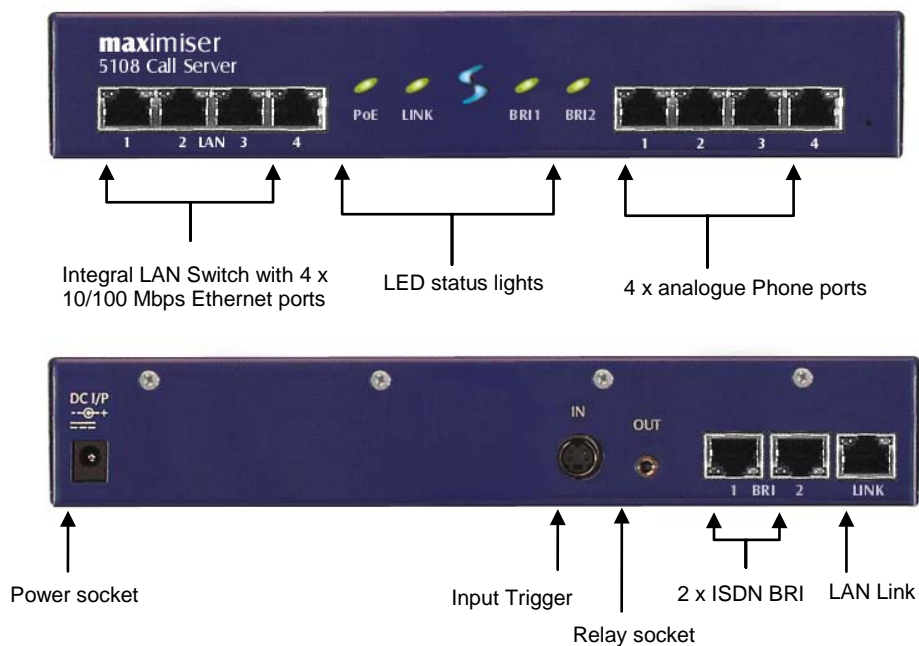
1. Power up the Call Server using the power supply provided.
2. On the 5100 and 5108 Call Servers the Disk Drive LED will flash to indicate a system check, and on the 4100 and 4140 Call Servers the NET LED will flash to indicate a system check.
3. The Call Server's IP address will be 192.168.0.1
4. The Call Server will search for a DHCP Server, if not found, the DHCP server functionality will be enabled. If the Call Server is connected to a network during power up and finds a DHCP Server its internal DHCP facility will be disabled.
5. This Call Server will become the Primary Call Server.
6. The SpliceCom LED will become solid when the unit is ready.

Please note : If a Call Server is connected to a network during power up and it detects that another Call Server exists on the network it will attempt to join this system. Ensure the Call Server is disconnected from the network during power up if this is not required. (Please refer to Connecting Multiple Call Servers from page 72 for further details.)

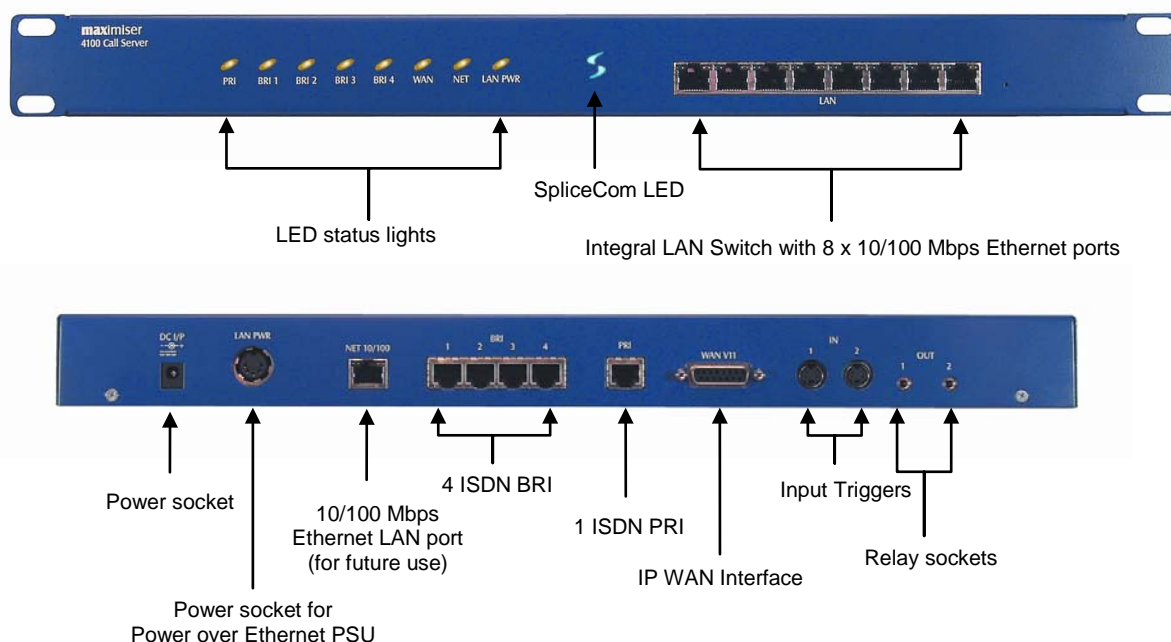## 5100 Call Server – Front and Rear view

5108 Call Server – Front and Rear view

Integral LAN Switch with 4 x 10/100 Mbps Ethernet ports

LED status lights

4 x analogue Phone ports

Power socket

Input Trigger
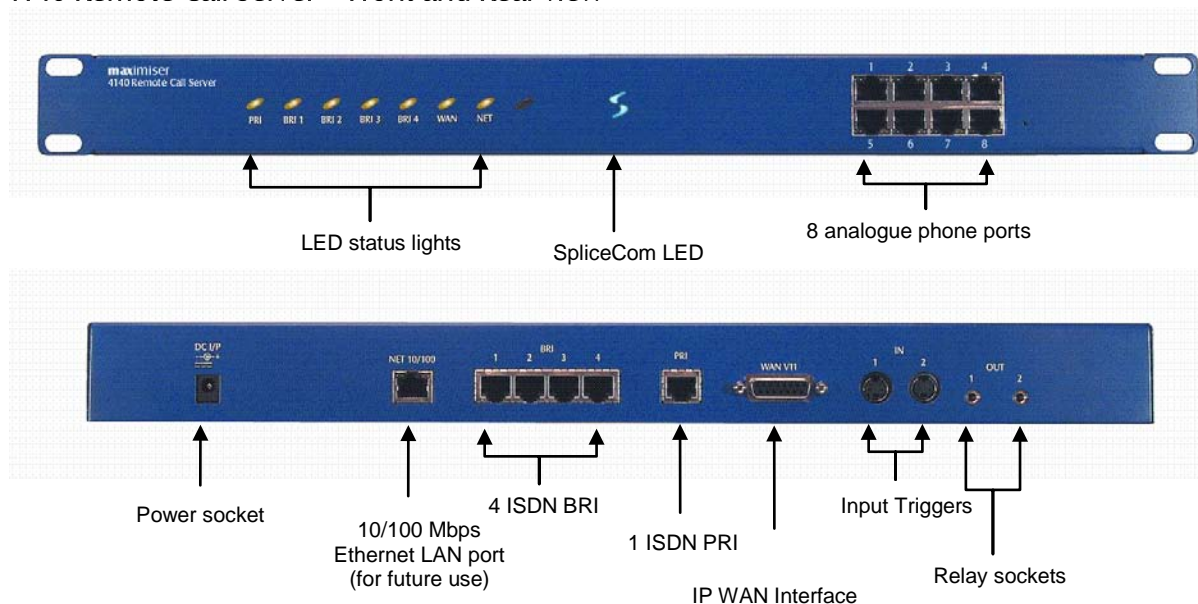
Relay socket

2 x ISDN BRI

LAN Link

Please note that the front view diagram is incorrect, the first LED is the LINK LED and the second is the DISK drive LED.

4100 Call Server – Front and Rear view

LED status lights

SpliceCom LED

Integral LAN Switch with 8 x 10/100 Mbps Ethernet ports

Power socket

Power socket for Power over Ethernet PSU

10/100 Mbps Ethernet LAN port (for future use)

4 ISDN BRI

1 ISDN PRI

IP WAN Interface

Input Triggers

Relay sockets

4140 Remote Call Server – Front and Rear view

LED status lights   SpliceCom LED   8 analogue phone ports

Power socket   4 ISDN BRI   Input Triggers

10/100 Mbps
Ethernet LAN port
(for future use)   1 ISDN PRI

IP WAN Interface   Relay sockets

Once the Call Server has been powered up the following should be noted:

- If the Call Server's DHCP Server functionality has been enabled the Call Server will give out an address range of 192.168.0.2 to 192.168.0.250, with 192.168.0.250 being the first address given out. (For further information on using DHCP please refer to page 29 for further details.)

- When connecting to the LAN ports on the front of the 5100, 5108 or 4100 Call Server straight or crossover cables can be used because each port automatically changes to MDI/MDX. When connecting to the LAN port at the back of the 4140 Remote Call Server a crossover cable is required.

- The ISDN Trunk ports available on a Call Server can be used to connect to the exchange or to another system. Please refer to page 159 for further information.

- The system operates in native mode at G.711 64k with echo cancellation. If the **max**imiser system is connected to the company LAN and that LAN is used to carry voice then that LAN must provide Quality of Service via LAN switches.

- The 5100 and 5108 Call Server and the 4140 Remote Call Server provide 16/4/8 analogue extension ports respectively. When either of these Call Servers is the first to be installed each port will be automatically assigned to a User named Extn2001, Extn2002 etc and given an extension numbers of 2001, 2002 etc. Please refer to page 48 for further information on using an analogue extension port.

- Please note that the first 8 analogue ports on the 5100 Call Server are activated, the remaining 8 ports will require a POTS licence to activate each port. Please refer to page 35 for further information on Licensing.

System Operation

Manager

The configuration of the **max**imiser system is controlled and stored on the LDAP database within the Call Server. Manager is the application that is used to view and configure this database.

Changes to the configuration made via Manager or by a User via their handset are stored in the RAM and copied regularly to the hard disk to be stored permanently. **It is important to perform an orderly shut down of the system before power down to ensure all changes are copied to the hard disk and so that the hard disk is shut down cleanly** - please refer to page 253 for further details.

Manager is a web based application which can be run from any Web browser (Internet Explorer v5 or higher, or Apple Safari v3.2 or higher) that can connect to the **max**imiser system.
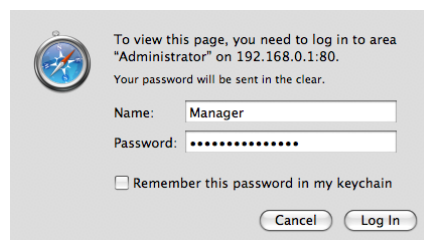
Running Manager

    1   Your PC must have a IP connection to the Call Server

    2   Via a web browser, open

        http://*ip address of Call Server*/manager   eg http://192.168.0.1/manager

    3   To view the system's configuration administration access is required. By default, full access is available by logging in when requested as follows:

        User Name = Manager
        Password = managerpassword

For security, it is advisable to at least change the password for this Administrator once access has been achieved. Please refer to the Setting Up Administrative Access section from page 31 for further details.

Working in Manager

On opening Manager a list of the configurable objects on the database is displayed in orange within the navigation pane on the left hand side (please refer to the Manageable Objects section from page 18 for a description of each object). A single click on one of the objects will display a list of the relevant entries in the database. The User's list is displayed by default.
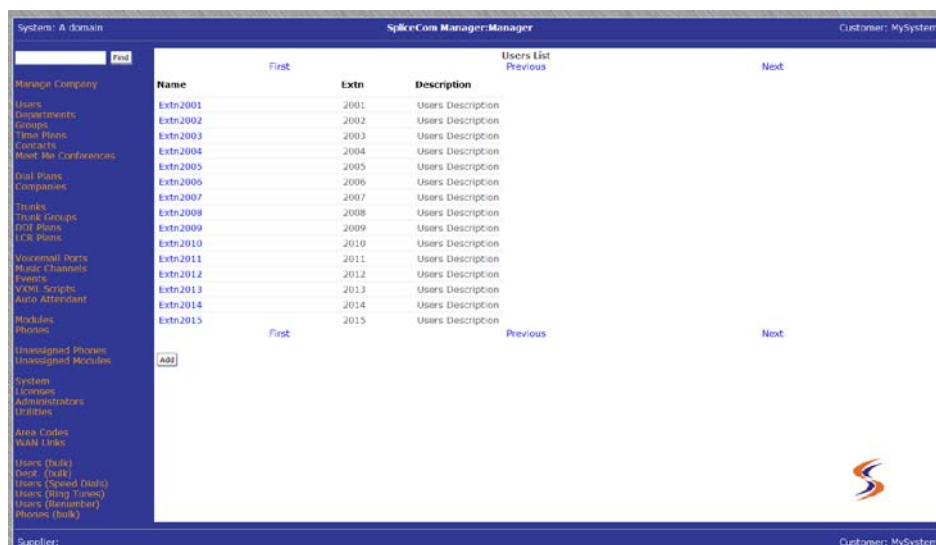
The following options will assist you to navigate and configure the database:

Navigate within a list of entries

First   Display the next page of entries

Previous   Display the previous page of entries

Next   Display the first page of entries

[Find box]   When a list of entries is displayed the Search box will be available at the top of the navigation pane. Enter the specific entry you wish to find in the Search box and then select Find or press Enter. The entry required will appear at the top of the list. If the Search box is empty, the Find button will take you to the top of the list.

[Add]   Create a new entry. The new entry will appear at the bottom of the list and will be called, for example, newDepartment, newUser, newLicence etc. Select this entry to configure the new item.

Configure an entry

Extn2001   To open, view and edit an entry click on the blue link

[Update]   To save the changes and return to the previous screen

[Apply]   To save the changes and remain in the current entry.

[Cancel]   To return to the previous screen. If changes have been made these will not be saved.

[Delete]   To delete the entry currently displayed. A confirmation message will be received.

[Clear]   To delete the current entry within the field

Telephony	An entry may consist of more than one page and the names of these pages are display at the top of the screen. Click on the page name to access that page. Use the Apply button to save a change in one page before moving to another page.
Dial Plan	When a field name is displayed in blue the entry for this field can be selected from a list. Click on the field name and select the new entry required. The Cancel button will return you to the previous screen. The Clear button will delete the entry and leave the field empty.
Standard	If an entry for a field is orange, clicking on this link will display the configuration form for this particular entry. The Cancel button will return you to the previous screen.

Configuring System Details

The System Details form of the configuration is used to enter information to identify the system such as the customer's name, the supplier's name and any other text that will help to distinguish the system. If licences have been purchased the text entered must match that provided with the licences. For further information please refer to the Installing Licences section from page 35.

1 In Manager, click on System
2 The System Details form will be displayed.
3 Enter a name for your system in the System Name field. Do not use punctuation in this field. This field is useful to identify the system for troubleshooting purposes. If you have purchased licences enter your given System Name. (This information will appear on the right hand side of the Manager screen at the top and bottom once this entry has been updated.)
4 The Supplier field is provided purely for information unless licences have been purchased. Enter the Supplier Name provided with the licences. (This information will appear on the left hand side of the Manager screen at the bottom once this entry has been updated.)
5 The Description field is provided purely for information to help you identify the system. This information will appear on the left hand side of the Manager screen at the top.
6 The System Locale field will determine the language used for voicemail, ring sequences and Caller Display.
7 Select Update or Apply when ready.

System Details	
System Name	Craddock and Shar Ltd
Supplier	Golden Telecoms
Description	A domain
Locale	eng
Call Logging Password	

For information on the use of the Call Logging Password please refer to the Call Logging section from page 27.

For information on the use of the Diagnostic Engineering Password, Web Password and Maintainer Password please refer to the System Access section on page 251.

Call Status

```
127.0.0.1
CallServer 00-07-d9-00-07-0b 4100 VCC38
```

Source	Source	Target	Target	Connected To	Connected To	Total	Now	#
Extn2002	2002	Extn2001	2001	Extn2001	2001	0:48	0:43	1
Extn2016	2016	Extn2018	2018	Extn2018	2018	0:27	0:21	2

Max calls (2)

Tue, 16 Jan 2007 16:00:50 +0000

Call Status can be accessed in Manager by selecting Utilities and then Call Status. This application allows all calls on the system to be viewed.

Source: Name and extension number of the User making the call

Target: The number dialled by the Source.   The User or Contact name will be displayed if matched to the number dialled

Connected to: The number the Source was actually connected to. The User or Contact name will be displayed if matched to the number dialled

Call Logging

Call information to be used by third party call logging applications can be obtained from the system via Telnet to port 4001.

Please refer to the Call Logging Format document available from Marketing for details on the description of each field.

Changing the Call Logging Password

The Call Logging Password is configurable within Manager and can be used to protect access to the call logging data.
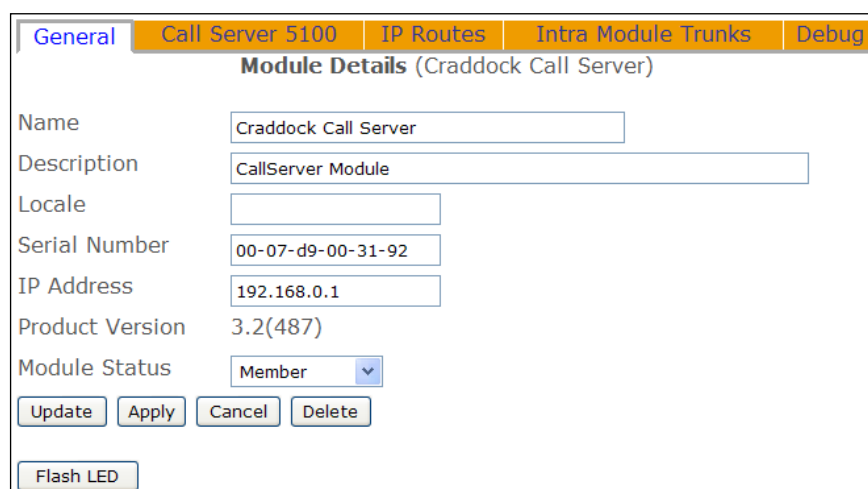
1   In Manager select System
2   In the Call Logging Password field enter the password required.
3   Select Update or Apply when ready.

No logging output will occur without the Call Logging password being received (new line terminated). The password must be transmitted in one block/packet.

Configuring the Call Server

The Call Server can be configured within Manager by selecting Modules and then the Call Server. The Module Details form for that Call Server will appear.

The Call Server's Name can be changed to help identify the Call Server. The module's hardware ID (Serial Number) is displayed together with its IP address and the current version of software being run on this Call Server.

General	Call Server 5100	IP Routes	Intra Module Trunks	Debug

Module Details (Craddock Call Server)

Name	Craddock Call Server
Description	CallServer Module
Locale	
Serial Number	00-07-d9-00-31-92
IP Address	192.168.0.1
Product Version	3.2(487)
Module Status	Member

[Update] [Apply] [Cancel] [Delete]

[Flash LED]

The Flash LED button will instruct the Call Server's SpliceCom LED to flash 20 times. This feature is useful when multiple Call Servers are being used on a system and will identify the physical Call Server from its configuration form. For information on working with multiple Call Servers please refer to page 72.

For further information on upgrading the software on a Call Server please refer to Upgrading a System from page 258. For further information on using the Module Status field please refer to Connecting Mulitple Call Servers on page 72.

Setting the IP Address of the Call Server

By default, a Call Server will have an IP address of 192.168.0.1. To change this address the following configuration can be made:

1   In Manager, select Modules
2   From the Modules list select the Call Server required
3   The Module Details form for that Call Server will be displayed.
4   Within the IP Address field enter the IP Address to be used by the Call Server.
5   Select Update or Apply when ready.
6   Reboot the Call Server (please refer to page 255 for information on how to reboot a Call Server.)

Please note:
- If the Call Server has DHCP enabled (as described below) and this IP address is not on the same network as the DHCP Start and End Address DHCP will not be operational.
- If the Call Server has DHCP disabled the DHCP Base Address and Base Mask must be configured to match the network that the Call Server is on.

A list of each module's IP address can be viewed in Manager by selecting Utilities and then IP Addresses.

31 users/devices have registered to this Call Server since last reboot.

IP Address	Port	Name found for address	Hardware Id
192.168.0.1		Modules.Craddock Call Server.BRI1	SpliceGateway 00-07-d9-00-18-23
192.168.0.247	5001	Users.Extn2012	SplicePOTS15 00-07-d9-00-04-95
192.168.0.248	5001	Users.Amy Stewart	SplicePhone 00-07-d9-00-18-f7
192.168.0.249	5001	Users.Sebastian Falks	SplicePhone 00-07-d9-00-15-43

13:06:04 Tuesday, 4th April 2006

Using DHCP
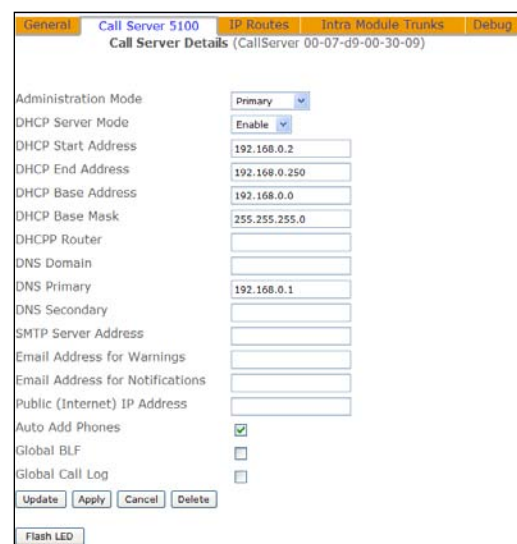
During power up a Call Server will search for a DHCP Server, if not found, its internal DHCP server functionality will be enabled and will give out an address range of 192.168.0.1 to 192.168.0.250. If the Call Server finds a DHCP Server the DHCP facility will be disabled.

To change this default configuration the following changes can be made.

1   In Manager, select Modules
2   From the Modules List select the Call Server.
3   The Module Details form for that Call Server will be displayed.
4   At the top of the form select the Call Server page. The Call Server Details form will appear.
5   From the DHCP Server Mode list the following options can be selected:

a   Enable – this is the default entry. The Call Server will act as the DHCP server using the IP Address range specified in the DHCP Start Address and DHCP End Address fields.

Please note: If another DHCP Server is detected on the network and the Call Server is rebooted it will continue to use 192.168.0.1or it's previously configured IP Address and disable its own DHCP Server which means it will no longer give out IP addresses.

If the DHCP Server mode is re-enabled having previously been disabled, the Call Server will firstly check whether another DHCP Server exists on the network. Only if it does not receive an IP address will it become a DHCP Server.

b   Disable – DHCP is disabled; the Call Server will use the IP address entered in the IP Address field within the General page. **(This is the recommended option if DHCP is being provided from another vendor)**

c   Client – the Call Server will act as the DHCP client obtaining an IP address from a DHCP Server on the network.

6   The DHCP Start Address and DHCP End Address fields specify the IP address range for the system when the DHCP Server Mode is set to Enable. (Please note that these fields are irrelevant if the DHCP Server Mode is set to Disabled.)

7   Select Update or Apply when ready.
8   Reboot the Call Server (please refer to page 255 for information on how to reboot a Call Server.)

Please note:
- Using the default configuration 192.168.0.250 will be the first IP address to be given out, followed by 192.168.0.249 etc.
- On initial connection a PCS 580, 570, 560, 410, 400 and 100 will obtain its IP address from a DHCP server.
- On initial connection a Phone Module will obtain its IP address from a DHCP server.

System Time

The System configuration form displays the current date and time used by the system. This setting can be changed as follows:

1   In Manager, select Utilities
2   From the Commands list select Set Time
3   Enter the date and time required
4   Select Update when ready.
5   This information will be sent and updated on all handsets on the system

Please note:
- The system will take into account British Summer Time.
- If multiply Call Servers are being used on one system the above time will be the time used by the Call Server whose Manager is being viewed. Each Call Server can have a different date and time.   This information will be sent and updated on all handsets registered to the specific Call Server.

A Time Zone can be set when the time displayed in, for example, call logging information needs to be identified with its Time Zone. This is useful when, for example, one maximiser system with multiple Call Servers is working across multiple countries or states and the time of a call needs to be identified with its correct Time Zone. (For further information on working with multiple Call Servers please refer to page 72.)

Each Call Server can have a different Time Zone and is configured as follows:

1   Open Manager on the Call Server to be configured
2   Select Utilities
3   Under Commands select Set Time
4   From the Time Zone list box select the option required
5   Select Update when ready.
6   Reboot the Call Server (please refer to page 255 for further information)

Please note: Setnet can be used to set the time zone for a specific Phone Module. For further information please refer to the Using Telnet section from page 270.

Setting up Administrative Access

Default Administrator

By default administration access to the database via Manager is available by logging in as "Manager" using the password "managerpassword". This Administrator has full access to the database.

It is recommended that at least the Password be changed to ensure security of the system.

1   In Manager, select Administrators
2   The Administrator List will be displayed
3   Select Manager
4   In the Password field enter the password required
5   Select Update or Apply when ready

Please note: This default Administrator, Manager, gives Authorised Access to new Administrators as described below. It is therefore recommended that Manager is not deleted.

Additional Administrators

Additional administrators can also be created to enable staff to log on to Manager but have limited access to the configuration. A new administrator can be created as follows.

Creating a new Administrator

1   In Manager, select Administrators
2   The Administrator List will be displayed
3   Select the Add button
4   Enter the name required for the new Administrator
5   Enter a description if required to further identify this Administrator
6   Enter the password to be used by this Administrator
7   Tick Show Enabled. This will enable the use of the Show tick boxes below. If not ticked this Administrator will have full access to the database (dependent on the rights and access allowed).
8   Tick the relevant box beside the section of the database to which this administrator is to have access eg Show Users, Show Departments etc
9   If you wish this Administrator to be able to create and/or delete entries within the section of the database selected above tick the relevant box, eg Add/Delete Users, Add/Delete Departments. This option will display the Add button within a list and the Delete button within a specific entry. (Please note that the Show option allows administrators to amend an entry even though Add/Delete is not selected (dependent on the rights and access allowed).)
10 Select Apply

11   If you wish the administrator to be shown a specific page of the database on opening Manager from the Default Page list box select the page required, eg Contacts.

12   Select Apply

Assigning rights to the Administrator

13   Within the Rights section at the bottom of the page select the Add Right button. A new entry will be displayed providing full rights to the database dependent on the options selected above.

14   Select Update when ready.

Authorising the Administrator

15   From the Administrator List select Manager

16   Within the Authorised Access section select the Add Access button

17   Select the Administrator field

18   From the Select Administrator list select the administrator created above. (This entry will provide full rights to the database dependent on the options selected within the Administrators configuration form.)

19   Select Update or Apply when ready.

View the configuration logged on as the new Administrator

20   Open Manager via your web browser.
21   In the User Name field enter the name of the new
      Administrator and enter this Administrator's password.
22   The Administrator logged in will only have access to the
      sections of the database configured above.

Please note: as the default Administrator, Manager, gives Authorised Access to new Administrators it
recommended that Manager is not deleted.

Creating an Administrator for PCS 60 Operator Console Mode

When a User is using PCS 60 in Operator Console Mode the operator can also be given the ability to
change a User's DND and Forwarding settings via their Favourites. To enable this facility the operator will
require an Administrator name and password.

1   Follow steps 1 to 7 above to create an new Administrator.
2   Select the Show Users option. This is the minimum option required for this facility to be available.
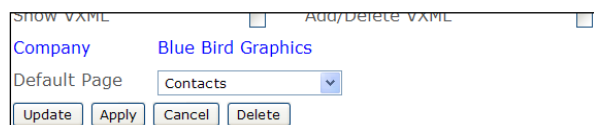

3   Follow steps 13 to 19 above to enter the Rights and
     Access levels for this Administrator.
4   Within the PCS 60 application enter the Administrator
     Name and Password in the PCS 60 Preferences
     dialogue box.

For further information on please refer to the PCS 60 Operator Console Mode section on page 68 and to
the PCS 60 User Manual.

Creating an Administrator for a Company

If the **max**imiser system is using the Companies feature you may wish an Administrator only to have access to the entries relating to a specific Company. In this case within the required Administrator's configuration form select the Company field and select the required Company from the Select Company list.

For further information on Companies please refer to page 155.

Using Auto Add Phones

This feature determines whether phones (analogue or IP) will be able to automatically register with the system.

If turned on (default), each Phone Module and IP Phone will be able to automatically register with the Call Server, each port and phone will be listed in the configuration and a User automatically created.

If turned off, each Phone Module and IP Phone must be manually registered on the system

Turn On/Off Auto Add Phones

1   In Manager, select Modules
2   Select the Call Server
3   Select the Call Server page.
4   In the Auto Add Phones check box tick to turn on or untick to turn off this feature.

Manually register a Phone Module during installation

If the Auto Add Phones feature is turned off a Phone Module must be manually registered during installation as follows:

1   In Manager select Unassigned Modules
2   The Phone Module will be displayed together with its MAC address
3   Select this link
4   The Module Status field will display Available
5   From the Module Status list box select Member
6   Select Update or Apply when ready
7   This unit will now be listed under Modules

The same method is used when installing a Trunk Module.

Please refer to the Connecting a Phone Module section from page 39 for further details.

Manually register a PCS 580/570/560/410/400/100 during installation

If the Auto Add Phones feature is turned off an IP phone must be manually registered during installation as follows:

1   In Manager select Unassigned Phones
2   The MAC address of the PCS will identify the unit connected. Click on the unit required
3   From the Status list box select Member
4   Select Update when ready
5   The unit will now appear under Phones

Please refer to the Connecting a PCS section from page 51 for further details.

Manually register a PCS 60 IP Phone or PCS 50 IP Phone during installation

If the Auto Add Phones feature is turned off a PCS 60/50 IP phone must be manually registered during installation as follows:

1   In Manager select Unassigned Phones
2   The hardware address of the PC will identify the PCS 60 to be connected. Click on the entry required.
3   From the Status list box select Member
4   Select Update when ready
5   The PCS 60/50 will now appear under Phones, identified by the hardware address of the PC

Please refer to the Connecting a PCS section from page 51 for further details.

Installing Licences

Certain functionality of the **max**imiser system requires the purchase and installation of the appropriate licence in order to activate the relevant facility.

Each User created on the system requires a licence. There are two types of User licences either an IPUser licence or an IPVirtualUser licence. An IPUser licence will be required to create a User who will be assigned to a physical telephone. By default the **max**imiser system is supplied with one IPUser licence which can be used for testing purposes. When a Call Server or Phone Module providing analogue ports is installed this is automatically supplied with the required number of IPUser licences and the corresponding number of analogue phone Users automatically created can utilize the system without any additional configuration. If further Users are to be created when connecting, for example, a PCS 60 (IP Phone), PCS 50 (IP Phone), PCS 580, 570, 560, 410/400 or 100 additional IPUser licences can be purchased. An IPUser licence will also be required to create a hot desking User as this User will be logging on to a physical handset.

An IPVirtualUser licence is required to create a User that requires PBX functionality but does not require a physical telephone.   For example, where a company offers virtual office facilities, voicemail and forwarding features are required but the customer will not be using a physical telephone. There is no limit to the number of IPVirtualUser licences that can be run on a Call Server.

An OperatorConsole licence is required for each User who will be using the PCS Operators Console application or the PCS 580/570/560/60 in Operator Console mode. Please note that this is a per-user

licence not a concurrent-usage licence. Please refer to the PCS 60 Operator Console Mode section from page 68 for further details.

To provide a User with voicemail facilities a MessageBox licence is required. On purchasing a 5100 Call Server or 4140 Remote Call Server 10 MessageBox licences are provided. A 5108 Call Server is provided with 5 MessageBox licences. A 4100 Call Server is provided with 30 MessageBox licences. Further MessageBox licences can be purchased to increase this number, up to 1,000 per system when using embedded voicemail or up to 100 on a standalone 4140 Remote Call Server and up to 50 on a standalone 5108 Call Server. MessageBox licences are also required to provide a Department with voicemail functionality. Please refer to the Voicemail section on page 187 for further details. (Please note: a list of which Users and Departments have been assigned MessageBox licences can be viewed in Manager by selecting System and then Mailbox Licences. Please refer to page 238 for further details.)

A Voice Processing Port licence will enable an additional port/simultaneous voicemail connection on a 5100 or 5108 Call Server. The relevant number of licences will also be required to run the voicemail application from a standalone Linux PC or server. One licence is necessary for each connection required. Please refer to the Working with Voicemail section from page 187 for further details.

To provide Auto Attendant/IVR functionality beyond the single layer provided as standard on **max**imiser, an ESPSession licence allows a single channel to be enabled. Enhanced Speech Processing (ESP) is a voicemail function and can run on the Call Server (up to eight licences), Remote Call Server (up to four licences) or a standalone Linux server (up to sixty licences). ESP is a VoiceXML based application and supports Text-to-Speech.

The 5100 Call Server is provided with four BRI ports making 8 ISDN channels available, 4 channels are enabled on installation however to enable an additional channel a Trunk licence is required. A Trunk licence is also required to activate a channel on the PRI ports. The 5108 Call Server provides two BRI ports giving 4 ISDN channels two of which are enabled on installation and Trunk licences are required to enable further channels. The 4100 Call Server and 4200 Trunk module are provided with four BRI ports providing 8 ISDN channels and a PRI port however to activate a PRI channel a Trunk licence is required. The Remote Call Server is provided with four BRI ports making 8 ISDN channels available, 4 channels are active on installation and a Trunk licence is required to activate a further channel. To activate a PRI channel (up to 15 channels) on a Remote Call Server a Trunk licence is also required. To use DPNSS, SIP or H.323 IP Trunk services a Trunk licence is necessary for each concurrent channel required.

The 5100 Call Server is provided with 16 analogue POTS ports of which 8 are activated on installation. To activate the remaining 8 ports a POTS licence is required for each port. Please refer to the Connecting an Analogue Extension section from page 48 for further information.

A 5108Plus licence will increase the number of users supported by a 5108 Call Server to 12. The maximum number of analogue users stays at four however the IP users can increase to 8, or in an IP only environment all 12 users could be IP. Note that the additional users will also need an IPUser Licence.

A VoiceCompression licence allows a further one channel of voice compression to be enabled per Voice Compression Module, giving a maximum of 32 compression channels per module

In summary, the licences available on a **max**imiser system are as follows:

5108Plus	Increases the number of users supported by a 5108 Call Server to 12. The maximum number of analogue users stays at four however the IP users can increase to 8, or in an IP only environment all 12 users could be IP. Note that the additional users will also need an IPUser Licence.
ESPSession	Enables a single Enhanced Speech Processing channel on the on-board or off-board voicemail application to facilitate the use of the Extended Attendant, VXML Scripts and Ring Back when Free. One licence per channel.
IPUser	Enables the ability to create a User that is assigned to a physical extension. One licence per User.
IPVirtualUser	Enables the ability for a User, which is not assigned to a physical extension, to be active on the system. One licence per virtual User.
MessageBox	Enables voicemail functionality for a User or Department. One licence per User or Department.
OCS	Enables the use of the interface, written by the Spider Group, between the **max**imiser and Microsoft Office Communications Server. Full instructions will be given on the purchase of this service.
OperatorConsole	Enables the use of the PCS Operators Console application or PCS 580/570/560/60 Operator Console mode. One licence per User.
POTS	Enables the use of an analogue extension port on the 5100 Call Server. One per port.
TotalControl	For use with the Total Control service. Full instructions will be given on the purchase of this service.
Trunk	Enables the use of an additional BRI or PRI channel, or utilize SIP, H.323 or DPNSS. One licence per channel.
VisionReport VisionRecording Vision Live VisionMobility	For use with Vision. Please refer to Vision Installation and Configuration Manual for further details.
VoiceCompression	Enables a compression channel on a Voice Compression module. One licence per channel.
Voice Processing Port	Enables an additional voicemail connection on a 5100 and 5108 Call Server and facilitate the use of the external voicemail application. One licence per port.

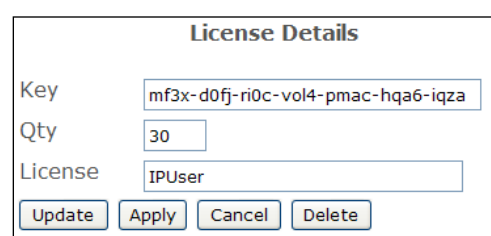## Configuring the System Name and Supplier Name

Licences are issued against a company name and the supplier's name therefore the System Name and Supplier Name provided with the licences must be entered accurately in the System configuration form. If this information is not entered correctly the licences will not activate.

Please refer to the Configuring System Details section from page 26 for further details.

Enter the Licences on the system

Each licence is provided with the type of licence, the quantity of licences purchased and a licence key. This information must be correctly entered; otherwise the licences will not be operational.

1   In Manager, select Licences
2   The Licences List will appear.
3   Click on the Add button
4   In the Key field enter the alpha-numeric code provided
5   In the Qty field enter the number of licences purchased
6   In the Licence field enter the type of licence purchased, eg IPUser, MessageBox
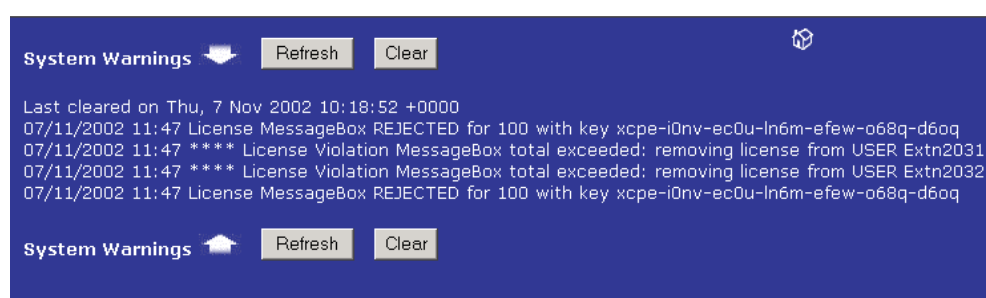7   Select Update or Apply when ready

License Details

Key	mf3x-d0fj-ri0c-vol4-pmac-hqa6-iqza
Qty	30
License	IPUser

[Update] [Apply] [Cancel] [Delete]

If a licence has not been entered correctly and not authenticated error messages will appear in the Warnings screen which can be accessed as follows:

1   In Manager, click on Utilities
2   Under the Diagnostics list select Warnings
3   Any error messages relating to licences will appear in this screen. No error messages indicate that the licences have been accepted and authenticated correctly.

System Warnings   [Refresh] [Clear]

Last cleared on Thu, 7 Nov 2002 10:18:52 +0000
07/11/2002 11:47 License MessageBox REJECTED for 100 with key xcpe-i0nv-ec0u-ln6m-efew-o68q-d6oq
07/11/2002 11:47 **** License Violation MessageBox total exceeded: removing license from USER Extn2031
07/11/2002 11:47 **** License Violation MessageBox total exceeded: removing license from USER Extn2032
07/11/2002 11:47 License MessageBox REJECTED for 100 with key xcpe-i0nv-ec0u-ln6m-efew-o68q-d6oq

System Warnings   [Refresh] [Clear]

Please note that licences can be downloaded directly from SpliceCom by using the Download Licences option. You will require an internet connection and you will need to know your Forums registered email address and password. Please refer to Support for further assistance.

View the current licence status

A list of the licences installed on the system together with how many of these licences have been used is available by opening:

http://<ip address of call server>/manager/licences.php

eg http://192.168.0.1/manager/licences.php

Licence Count:

Licence Type	Available	Used *
IPUser	34	29
MessageBox	30	30
OperatorConsole	1	0
voicemailPort	4	4

* Only the Primary counts the licences used.
Note: Adding a new/duff licence will trigger a recount.

Modules affecting licence count in system:

#	IP Address	Name	Type	Version	Last Seen
1	192.168.6.70	Craddock Call Server	5100	3.2(1284)	0 days ago @ 16:06:30 Wednesday, 3rd March 2010
2	192.168.6.15	Phone Module 00-07-d9-00-31-20	5315	3.2(1057)	0 days ago @ 12:16:58 Wednesday, 3rd March 2010
3	192.168.0.251	VisionServer 00-26-4a-09-07-e0	20	1.0(33)	25 days ago @ 02:37:44 Saturday, 6th February 2010

18:05:55 Wednesday, 3rd March 2010

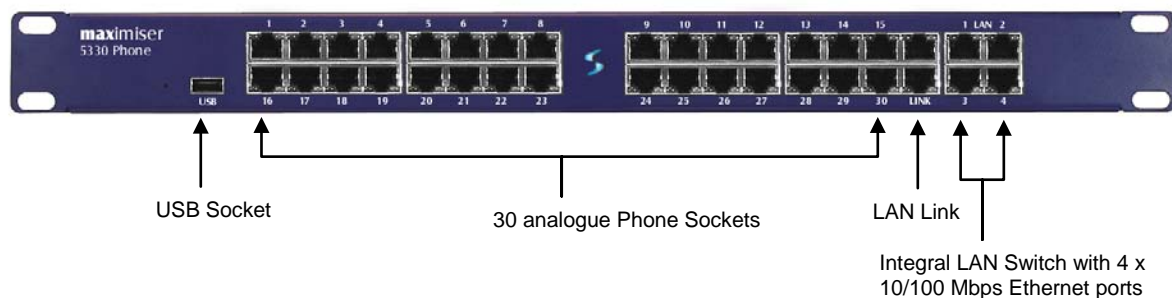# Connecting a Module

Connecting a Phone Module

The Phone module enables connection of analogue handsets, including PCS 5/10s, to the system. All configuration information is obtained from the Call Server and is stored in RAM.

1 Connect the Phone module to the system via a LAN port on the front of the Phone module. This can be either via the network or directly to a LAN port on the Call Server. (A straight LAN cable should be used when connecting the Phone Module to third party LAN switches.)
2 Power up the Phone Module.
3 The module will request an IP address from a DHCP server.   It will make 4 attempts and then back off for a period before trying again. The back off period will extend for each failed attempt.
4 Having obtained an IP address the module broadcasts to find a Call Server to provide its configuration and become active.
5 The SpliceCom LED will become solid when the unit is ready.
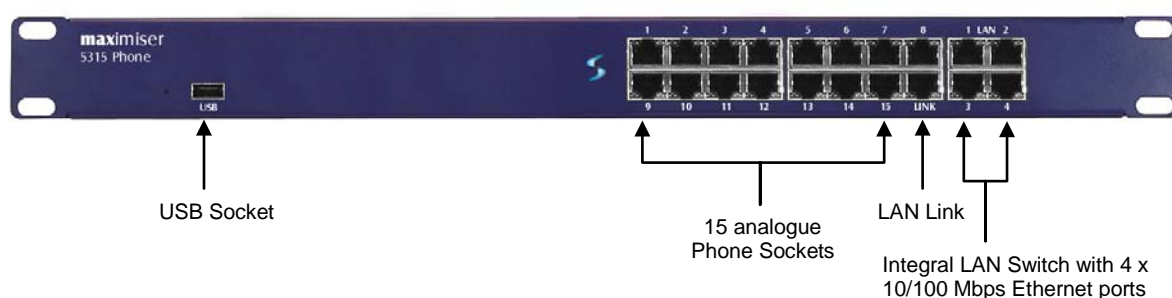6 If the Auto Add Phones feature is turned on the Phone Module will automatically register with the system

   If the Auto Add Phones feature is turned off the Phone Module must be manually registered as follows:

   a In Manager select Unassigned Modules
   b The Phone Module will be displayed together with its MAC address
   c Select this link
   d The Module Status field will display Available
   e From the Module Status list box select Member
   f Select Update or Apply when ready
   g This unit will now be listed under Modules

7 Each analogue handset previously connected will ring once to confirm communication with the system
8 Connect each analogue handset to a port on the Phone module via a master dongle or RJ45 connection if supplied. For further details on connecting an analogue handset please refer to page 48.
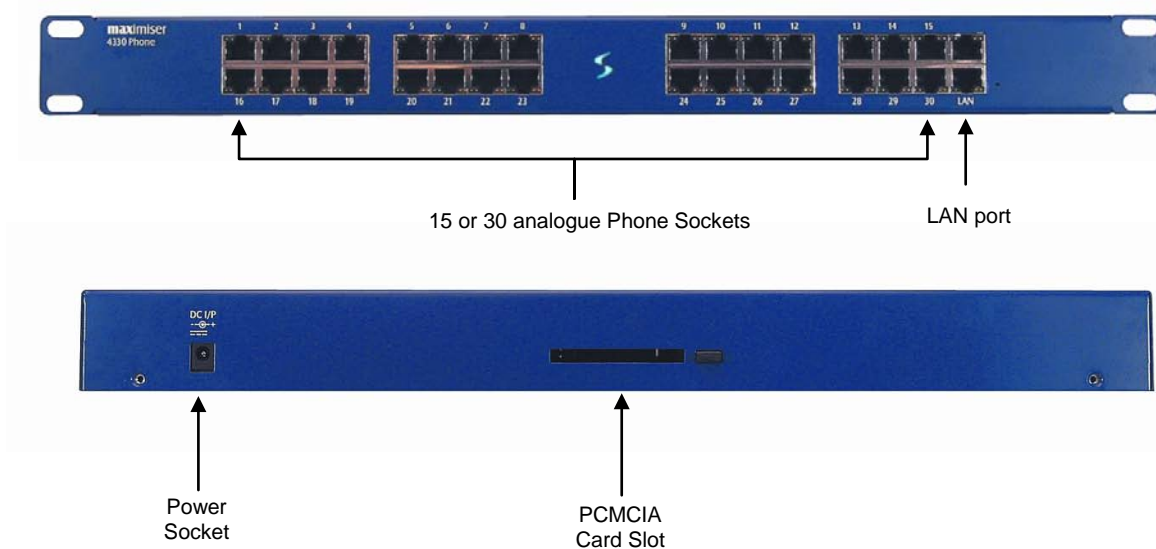
5330 Front View

USB Socket

30 analogue Phone Sockets

LAN Link

Integral LAN Switch with 4 x
10/100 Mbps Ethernet ports

5315 Front View

USB Socket

15 analogue
Phone Sockets

LAN Link

Integral LAN Switch with 4 x
10/100 Mbps Ethernet ports

5315/30 Rear View

Power socket

4300 Front and Rear View

15 or 30 analogue Phone Sockets

LAN port

Power
Socket

PCMCIA
Card Slot

The Phone Module controls:
- Ringing of the telephones at the requested cadence
- Presentation of the CLI
- Recognition of off-hook state
- Recognition of dialed DTMF tones
- Presentation of DTMF for fax servers
- The left hand LED on a port displays when a phone is ringing
- The right hand LED on a port displays when a phone is in use
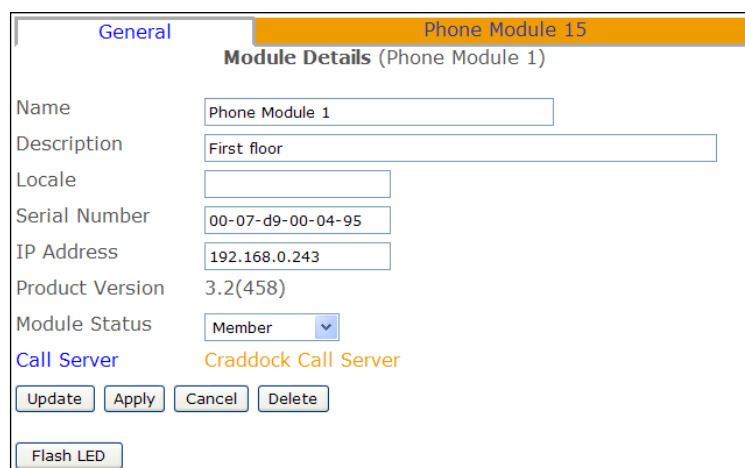
Please note:
- Anti static precautions are advised when handling, inserting or removing PCMCIA cards with respect to the 4300 Phone modules.
- Calls placed on hold will be automatically taken off hold after 10 minutes
- On a reboot of the Phone Module each analogue handset will ring once to confirm communication with system and display the number of new voicemail messages waiting for the User assigned to that phone (if voicemail enabled).

Configuring a Phone Module

1. In Manager select Modules
2. The Phone Module will be displayed together with its MAC address
3. Select this link to view and configure this module.
4. For example, you may wish to edit the Name and Description of this module to give an identity and location for future reference.
5. Select Update or Apply when ready.

The Call Server field indicates to which Call Server the Phone Module is registered. The Call Server is responsible for replicating relevant configuration information to the Phone Module. In turn the Phone Module will store in RAM configuration information relevant to its functionality. The Phone Module will also pass back configuration changes, eg User configuration, to the Call Server.

The Flash LED button will instruct the Phone Module's SpliceCom LED to flash. This feature is useful when multiple Phone Modules are being used on a system and will identify the physical Phone Module from its corresponding configuration form.
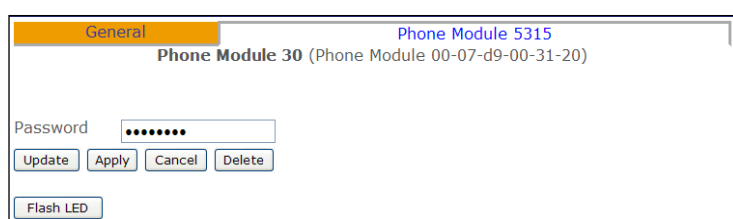
The Product Version field will display which version of software is currently running on the Phone Module. For information on upgrading this software please refer to page 260.

Each port on the Phone Module can now be configured independently, see page 77 for further details.

Changing the Password

When accessing a Phone Module via Telnet, when upgrading for example, the module's password is entered on connection. By default this is set to 7388. However this password can be changed as follows:

1   In Manager select Modules
2   Select the Phone Module required
3   Select the Phone Module page
4   In the Password field enter the password required
5   Select Update or Apply when ready.
6   Reboot the Phone Module

Extension Numbering

Once a Phone Module has registered with the system each port on the Phone Module, eg Port01, Port02 etc, will be automatically assigned to a User eg Extn2017, Extn2018 etc, and given an extension number, eg 2017, 2018 etc.

Extension numbering starts with the next available number, therefore if, for example, the existing extension number range on a system is 2001-2016, and a 5130 Phone Module is installed the Users automatically created will be given an extension number within the range of 2017-2046.
Each User can be amended to change this default configuration if required, see page 80 for further details.

To change the extension number range automatically used when a Phone Module registers with a system, eg 6001-6030, the following steps can be taken:

1   Before the Phone Module is connected to the system create a new User with an extension number one before the start of the range required, eg 6000
2   Connect the Phone Module as described on page 39.
3   The new Users created will be given an extension number within the new extension number range, eg 6001-6030.
4   The User created in step 1 can be deleted if no longer required.

Setting the IP Address

When connecting a Phone Module to the system for the first time a DHCP server is required to give the Phone Module an IP address. This address can be viewed within the Phone Module's configuration form as described on page 40. However once connected the Phone Module can be given a static address as follows:

1   Within the Command prompt enter

     telnet <ip address of Phone Module>   eg telnet 192.168.0.250

     then press Enter

2   You will be prompted for a Username, enter *diag*, then press Enter
3   You will be prompted for the password for this Phone Module. By default this is *7388*. (The Password is set within the relevant Phone Module's configuration form.) Then press Enter.
4   Wait for a # to appear to confirm a connection has been made to the Phone Module
5   Enter *setnet* then press Enter
6   The current configuration will be displayed. You will be asked if you wish to change any of this configuration.
7   Enter *y* then press Enter
8   You will be asked how the IP address will be assigned.
9   Delete the *d* and enter *s* then press Enter
10   Follow the onscreen prompts entering the IP address you wish to give the Phone Module, eg 192.168.0.50, the netmask you wish to use, eg 255.255.255.0 , the IP address of the Gateway (Default Router), eg 192.168.0.1 and the IP address of the TFTP server which will be the IP address of the Call Server to provide software to the Phone Module during an upgrade, eg 192.168.0.1.
11   At the following prompts make any changes you require otherwise press Enter to leave the option unchanged.
12   You will be asked if you wish to save the changes. Enter *y* then press Enter
13   Telnet will display that the changes are being saved and *Done* will appear once finished.
14   Enter *poweroff* then press Enter and the Phone Module will reboot
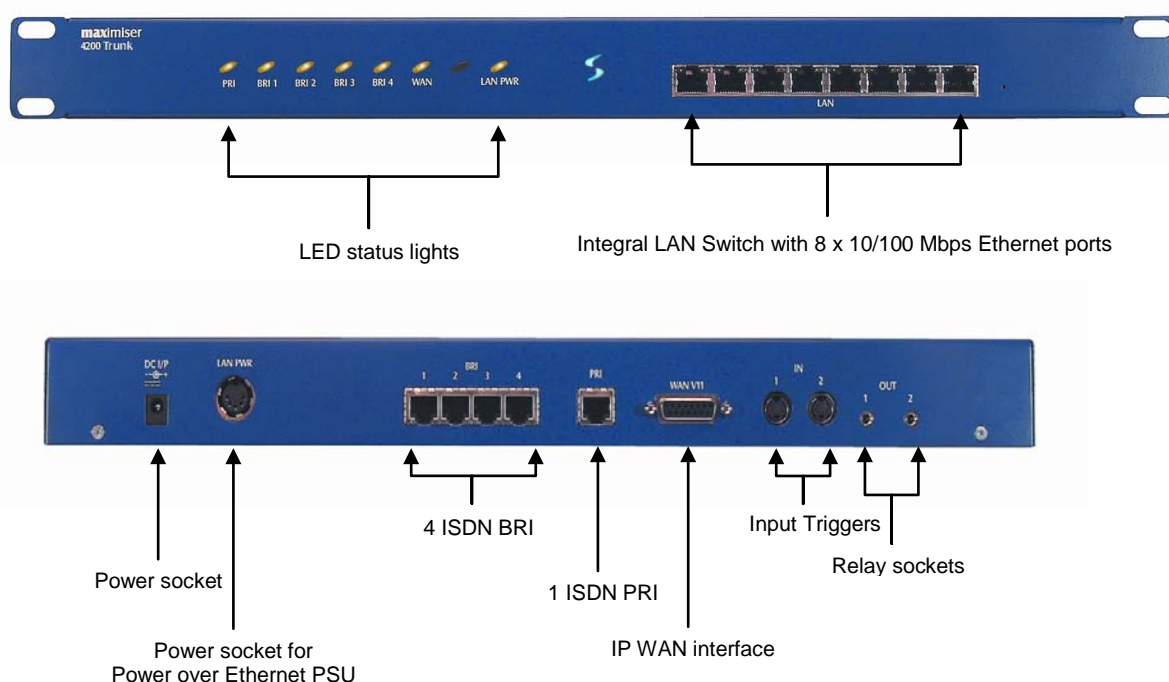
Please note:
- to return to using DHCP follow the same instructions above. Enter *d* when asked how the IP address will be assigned.
- at any prompt you may enter a *?* for help on that option or *q* to quit.
- for further information on working with Telnet please refer to page 270.
- a list of each module's IP address can be viewed in Manager by selecting Utilities and then IP Addresses.
- Setnet can be used to set the Time Zone to be used by a specific Phone Module

Connecting a Trunk Module

A Trunk Module can be connected to a system to increase the number of ports required, for example, to provide additional ISDN ports, an additional WAN port etc. A Trunk licence will be required to activate a PRI channel if required. Please refer to page 35 for further details.

1   Connect the Trunk Module to the system via one of LAN ports on the front of the Trunk Module. This can be either via the network or directly to a LAN port on the Call Server.
2   Power up the Trunk Module.

3   The module will request an IP address from a DHCP server.   It will make 4 attempts and then back
     off for a period before trying again. The back off period will extend for each failed attempt.
4   Having obtained an IP address the module broadcasts to find a Call Server to provide its
     configuration and become active.
5   The SpliceCom LED will become solid when the unit is ready.
6   In Manager select Unassigned Modules
7   The Trunk Module will be displayed together with its MAC address
8   Select this link
9   The Module Status field will display "Available"
10   From the Module Status list box select "Member"
11   Select Update or Apply when ready
12   This unit will now be listed under Modules

LED status lights

Integral LAN Switch with 8 x 10/100 Mbps Ethernet ports

4 ISDN BRI

Input Triggers

Power socket

1 ISDN PRI

Relay sockets

Power socket for
Power over Ethernet PSU

IP WAN interface

Configuring a Trunk Module

1   In Manager select Modules
2   The Trunk Module (SpliceGateway) will be displayed together with its MAC address
3   Select this link to view and configure this module.
4   For example, you may wish to edit the Name and Description of this module to give an identity and
     location for future reference.
5   Select Update or Apply when ready

The Call Server field indicates to which Call Server the Trunk Module is registered. The Call Server is
responsible for replicating relevant configuration information to the Trunk Module. In turn the Trunk
Module will store in RAM configuration information relevant to its functionality.

The Flash LED button will instruct the Trunk Module's SpliceCom LED to flash. This feature is useful when
multiple Trunk Modules are being used on a system and will identify the physical Trunk Module from its
corresponding configuration form.

The Product Version field will display which version of software is currently running on the Trunk Module. For information on upgrading this software please refer to page 260.

Setting the IP Address of the Trunk Module

When connecting a Trunk Module to the system for the first time a DHCP server is required to give the Trunk Module an IP address. This address can be viewed within the Trunk Module's configuration form as described above. However once connected the Trunk Module can be given a static address as follows:

1   In Manager select Modules
2   Select the Trunk Module (SpliceGateway)
3   Within the IP Address field enter the IP Address to be used by the Trunk Module.
4   Select Update or Apply when ready
5   Reboot the Trunk Module (please refer to page 255 for information on how to reboot a Trunk module.)

A list of each module's IP address can be viewed in Manager by selecting Utilities and then IP Addresses.

Each port on the Trunk Module can now be configured independently, please refer to Working with Trunks on page 159 and Creating a WAN Link on page 247 for further details:

Connecting a Voice Compression Module

The 4400 Voice Compression Module provides 16 channels, of 8kbps, G.729a based compression and can be deployed to provide a higher density of calls between Call Servers over low speed WAN links. Additional channels of voice compression can be enabled via a licence – giving a maximum of 32 compression channels per module (please refer to the Licensing section on page 35 for further information). By using a standard LAN for interconnectivity between the Voice Compression Module and its associated Call Server, there is no limit to the number of compression channels that can be practically deployed. This allows multiple 4400 Voice Compression Modules to be used with single or multiple Call Servers, with the compression channels being dynamically allocated as, or when, required.

1   Connect the VCM to the system via the LAN port on the front of the VCM Module. This can be either via the network or directly to a LAN port on the Call Server.
2   Power up the VCM.
3   The module will request an IP address from a DHCP server.   It will make 4 attempts and then back off for a period before trying again. The back off period will extend for each failed attempt.
4   Having obtained an IP address the module broadcasts to find a Call Server to provide its configuration and become active.
5   The SpliceCom LED will become solid when the unit is ready.
6   In Manager, select Unassigned Modules
7   The VCM will be displayed together with its MAC address
8   Select this link
9   The Module Status field will display "Available"
10   From the Module Status list box select "Member"
11   Select Update or Apply when ready
12   This unit will now be listed under Modules

Power
socket

For information on installing a Voice Compression card please refer to page 47.

A Voice Compression Module or Card will required at both ends to provide compression for calls between two Call Servers.

Configuring a Voice Compression Module

1   In Manager select Modules
2   The VCM (SpliceCompressor) will be displayed together with its MAC address
3   Select this link to view and configure this module.
4   For example, you may wish to edit the Name and Description of this module to give an identity and location for future reference.
5   Select Update or Apply when ready

The Call Server field indicates to which Call Server the VCM is registered. The Call Server is responsible for replicating relevant configuration information to the VCM. In turn the VCM will store in RAM configuration information relevant to its functionality.

The Flash LED button will instruct the VCM's SpliceCom LED to flash. This feature is useful when multiple VCMs are being used on a system and will identify the physical VCM from its corresponding configuration form.

The Product Version field will display which version of software is currently running on the VCM. For information on upgrading this software please refer to page 260.

Setting the IP Address

When connecting a VCM to the system for the first time a DHCP server is required to give the VCM an IP address. This address can be viewed within the VCM's configuration form as described above. However once connected the Voice Compression Module can be given a static address as follows:
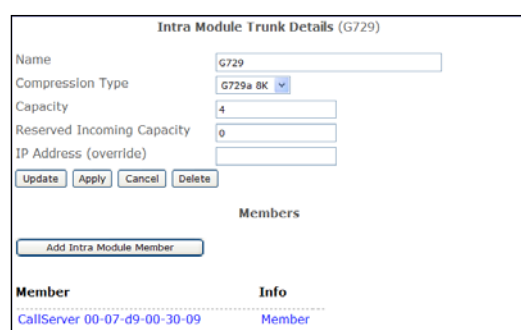
1   In Manager select Modules
2   Select the VCM (SpliceCompressor)
3   In the IP Address field enter the IP Address to be used by the VCM.
4   Select Update or Apply when ready.
5   Reboot the VCM (please refer to page 255 for information on how to reboot a VCM.)

A list of each module's IP address can be viewed in Manager by selecting Utilities and then IP Addresses.

Creating an Intramodule Trunk

In order to use the compression provided by a Voice Compression Module or Card (see below) the following should be configured. For calls between two Call Servers a Voice Compression Module or Voice Compression Card is required at both ends together with the following configuration.
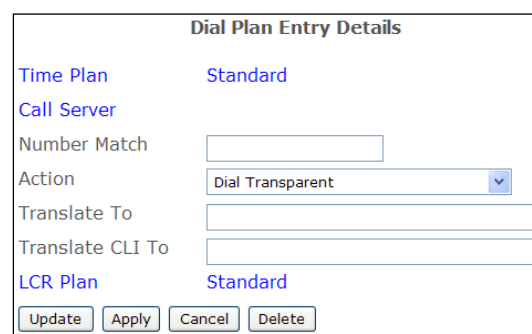
1   In Manager select Modules
2   Select the Call Server to which the VCM is connected or in which the card has been installed.
3   Select the Intra Module Trunks page
4   Click on Add Intra Module Trunk
5   In the Name field enter a name to identify this trunk
6   From the Compression Type list box select G729A8K
7   In the Capacity field enter the number of connections to be allowed on this link
8   Select Apply when ready
9   Select the Add Intra Module Member button and the list of Modules will appear
10 Select the Call Server at the other end
11 Repeat steps 11 to 12 until all the required Call Servers that have a VCM connected or containing a Voice Compression Card have been entered.
12 Select Update or Apply when ready.

If calls are to be made to a Call Server that does not contain a Voice Compression Card or does not have a Voice Compression Module connected it is recommended that the this Intramodule Trunk is also created.

If fax calls are to be made between two Call Servers where compression is supported a Dial Plan entry similar to the this example should be created within each fax machine's Dial Plan. The DialTransparent Action will allow calls to by-pass an Intramodule Trunk. A call made via this Dial Plan entry does not consume a compression channel. (For further information on Dial Plans please refer to page 114.)

Installing a Voice Compression Card

A Voice Compression Card supports G.729a to compress up to 8 voice calls between two Call Servers. A card must be fitted at both ends as follows:

4100 Call Server
1   Remove the lid
2   Place nylon standoffs supplied in mounting holes located by silk screen logos MTG14 and MTG15.
3   Fit VCC card to connector J12 and standoffs - take care to correctly align the pins.
4   Replace the lid.

4140 Remote Call Server
1   Remove the lid
2   Place nylon standoffs supplied in mounting holes located by silk screen logos MTG9 and MTG10.
3   Fit VCC card to connector J11 and standoffs - take care to correctly align the pins.
4   Replace the lid.

Connecting an Analogue Phone

Analogue extension ports are available on 5100 Call Server, 4140 Remote Call Server and 5300/4300 Phone Modules providing 16/8/15/30 ports respectively.

When a 5100 or 5108 Call Server, 4140 Remote Call Server or Phone Module is connected to the system 4/8/15/16/30 Users are created and named, for example, Extn2001, Extn2002 etc. Each User is assigned to an extension port, for example, Port 1, Port 2 etc, and given an extension number within the next available extension range, for example, 2001, 2002 etc. This default configuration can be changed as detailed within the Working with Users section from page 80.

To change the extension number range automatically used when a 5100 Call Server, 4140 Remote Call Server or Phone Module registers with a system, eg to 6001-6030, the following steps can be taken:

1   Before the Call Server/Phone Module is connected to the system create a new User with an extension number one before the start of the range required, eg 6000
2   Connect the Call Server/Phone Module as described from page 39.
3   The new Users created will be given an extension number within the new extension number range, eg 6001-6030.
4   The User created in step 1 can be deleted if no longer required.

To change the extension numbers for multiple Users after they have been created please refer to page 84.

A PCS 520, PCS 505, PCS 10, PCS 5 or an alternative analogue telephone can be connected to an analogue port via a master dongle or via a RJ45 connection if supplied.

Each port is provided with two LEDs to display the following activity on the port:

Left hand LED = displays when phone ringing
Right hand LED = displays when phone in use

When a 5100 or 4140 Call Server or Phone Module is rebooted each analogue handset will ring once to confirm communication with the system and will display the number of new voicemail messages waiting for the User assigned to that phone (if voicemail enabled).

Calls placed on hold will be automatically taken off hold after 10 minutes.

Analogue Ports on a 5100 Call Server

The first eight analogue ports on the 5100 Call Server are activated on installation. However ports 9-16 require a POTS licence to enable each port. Once the required number of POTS licences have been installed as described in the Installing Licences section from page 35 the following configuration is required:

1   In Manager select Phones
2   Select the relevant 5100 Call Server
3   Select the relevant analogue port – port09-port16
4   Remove the tick from the Unlicensed Port field
5   Select Update or Apply when ready.

For further information on configuring an analogue port please refer to page 77.

Each User assigned to ports 9-16 will also required configuration as follows:
1   In Manager select Phones
2   Select the relevant 5100 Call Server
3   Select the relevant analogue port – port09-port16
4   Select the entry displayed in the User field eg Extn2009
5   The User configuration form will be displayed
6   Remove the tick from the Ex Directory field (unless required)
7   Select Apply
8   Select the Licences page
9   Remove the tick from the Disable (lack of licence) field
10   Select Update or Apply when ready.

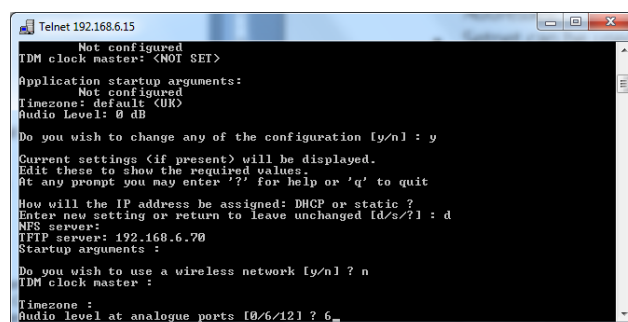For further information on User configuration please refer to the Working with Users section from page 80.

Increasing the volume on an analogue port

Phone Module
Setnet can be used to increase the volume given out on each analogue port on a Phone Module as follows:

1   Within the Command prompt enter

   telnet <ip address of Phone Module>   eg telnet 192.168.0.250

   then press Enter

2   You will be prompted for a Username, enter *diag*, then press Enter
3   You will be prompted for the password for this Phone Module. By default this is *7388*. (The Password is set within the relevant Phone Module's configuration form.) Then press Enter.
4   Wait for a hash (#) to appear to confirm a connection has been made to the Phone Module
5   Enter *setnet* then press Enter
6   The current configuration will be displayed. You will be asked if you wish to change any of this configuration.
7   Enter *y* then press Enter

8   Press Enter until the line "Audio Level" appears

9   Type in either 6 (recommended) or 12 depending on the volume level required.

10   Press Enter

11   You will be prompted to save this new configuration. Enter *y*.

12   The configuration will be saved.

13   Enter *poweroff* to reboot the phone module.

Call Server

To increase the volume output for the analogue ports on a Call Server a file called pots_level must be created within the /SpliceCom directory as follows:

1   Telnet on to the Call Server as described in the Root Access to a Call Server section from page 252

2   Enter *vi /SpliceCom/pots_level*

3   Press i to go to Insert mode

4   Type in either *6* (recommended) or *12*

5   Press Esc to exit Insert mode

6   Enter :wq to quit the file and save the changes.

7   Reboot the Call Server

Connecting a PCS 520

When connecting a PCS 520 to an analogue port the following should be considered:

Caller Display Type

The Default Caller Display Type for an analogue port will be suitable for use with the PCS 520 however to ensure that CLI information is displayed correctly and to allow the handset is to support Paging and Auto Answer the System Phone Caller Display Type should be used. For details on how to configure an Analogue Extension port please refer to page 77.

Default Dial Plan entries

The PCS 520 is supplied with 10 pre-programmed buttons which dial specific dial plan entries. Therefore the User assigned to the PCS 520 must be configured with a Dial Plan containing these default dial plan entries. For details on working with Dial Plans please refer to page 114 and, within this section, details on using the default dial plan entries from page 121.

Connecting a PCS 10

When connecting a PCS 10 to an analogue port the following should be considered:

Caller Display Type

The Default Caller Display Type for an analogue port uses the Caller Display Type required for use with the PCS 10 however if the handset is to support the Message Waiting light the PCS 10 Lamp Caller Display Type should be used. For details on how to configure an Analogue Extension port please refer to page 77.

Default Dial Plan entries

The PCS 10 is supplied with 10 pre-programmed buttons which dial specific dial plan entries. Therefore the User assigned to the PCS 10 must be configured with a Dial Plan containing these default dial plan entries. For details on working with Dial Plans please refer to page 114 and, within this section, details on using the default dial plan entries from page 121.

Connecting a PCS 505 or PCS 5

When connecting a PCS 505 or PCS 5 to an analogue port the following should be considered:

Caller Display Type

The Default Caller Display Type for an analogue port uses the Caller Display Type required for use with the PCS 505 or PCS 5 however if the handset is to support the Message Waiting light the Lamp Caller Display Type should be used. For details on how to configure an Analogue Extension port please refer to page 77.

Default Dial Plan entries

The PCS 505 and PCS 5 are supplied with a System Feature Guide displayed on the handset detailing the use of specific dial plan entries. Therefore if the User is going to utilize these dial plan entries the User must be configured with a Dial Plan containing these entries. For details on working with Dial Plans please refer to page 114 and, within this section, details on using the default dial plan entries from page 121.

Connecting a PCS IP Phone

Power over Ethernet considerations

Power to the PCS 580G, PCS 570G/570, PCS 560, PCS 410/400 and PCS 100 is provided by Power over Ethernet (PoE). This can be supplied by a third party PoE switch or by the following methods:

- PCS 580/570/560 – via one of the PoE LAN ports on a Call Server or Phone Module, or via a Single Terminal Ethernet Power Supply (STEPS)
- PCS 410/100 - via one of the PoE LAN ports on a Call server or Phone Module, or via a Single Terminal Ethernet Power Supply (STEPS)
- PCS 400 - via one of the PoE LAN ports on a Call server or Phone Module, or via a Single Terminal Ethernet Power Supply (STEPS). The PCS 400 can also obtain power via a PCS PSU when it is not practical to use PoE. Please note that if a PCS 400 is connected to a PoE LAN port on a 5100 or 5108 Call Server the PCMCIA card cannot be used.

Please note that if a STEPS is used with a PCS 580G or PCS 570G the LAN port (port 1) will not run at 1G.

PoE LAN ports are provided as follows:

5100 Call Server, 5108 Call Server and 5300 Phone Module – provides 4 x PoE LAN ports supporting the following combinations:
>	1 x PCS 580G or
>	1 x 410/400 or
>	up to 2 x PCS 570G/570 or
>	up to 4 x PCS 100/PCS 560 or
>	up to 1 x PCS 570 and 2 x PCS 100/PCS 560

4100 Call Server and 4200 Trunk Module – PoE is supplied to the LAN ports via the Power over Ethernet PSU connected to the LAN PWR port on the back of these modules. This will give power to the 2 spare pairs on each of the 8 Ethernet ports on front of these Call Servers supporting up to eight IP phones (PCS 580, 570, 560, 410/400, 100) in any combination.

(Please note that one PoE port cannot be used to provide PoE to ports on a hub or switch connected to this port.)

PCS 560/570/580

Power to a PCS 560/570/580 is supplied by Power over Ethernet (PoE). This can be provided either by a third party PoE switch or directly via a PoE LAN port available on a Call Server or 5300 Phone Module.

1   Connect the PCS 560/570/580 (via its LAN port numbered 1) to the system via the network or directly to a LAN port on the Call Server or 5300 Phone Module
2   The PCS 560/570/580 is a DHCP client and must obtain its IP address from a DHCP server.
3   An IPUser licence is required and must be installed before registration (See page 35 for further information.)
4   If the Auto Add Phones feature is turned on the Call Server will automatically register the PCS. (Please refer to page 34 for further details.)

    If the Auto Add Phones feature is turned off:
    a   In Manager select Unassigned Phones
    b   The MAC address of the PCS will identify the unit connected. Click on the unit required
    c   From the Status list box select Member
    d   Select Update when ready
    e   The unit will now appear under Phones

5   Once the system has registered the PCS a User will be automatically created, configured with the next available extension number.
6   The User's Initial Phone field displays the name of the PCS that is assigned to this User.

General	Details	Telephony	DND	Capability	Tunes	Speed Dials	Voicemail	Licenses

User General (Extn2017)

Name	Extn2017
Description	Users Description
Telephone Number	2017
Initial Phone	Craddock Call Server.PCS 00-07-d9-00-00-cb
Locale	
Company	
Current Home	Craddock Call Server

Please note that the phone's LAN port numbered 2 can be used to connect the User's PC, for example.

Each PCS 560/570/580 connected can be configured independently, see page 79 for further details.

Setting the Password

When using Telnet to, for example, upgrade the phone or change its IP Address. The phone's password is entered to authorise these changes. By default this is set to 7388 but this password can be changed as follows:

1. In Manager, from the Navigation pane, select Phones
2. Select the module that the phone is connected to, eg Call Server
3. The MAC address of the unit will identify the specific phone required
4. Select the phone required
5. In the Password field enter the alpha-numeric characters required
6. Select Update or Apply to save the changes.

SpliceCom IP Phone	General
SpliceCom IP Phone Details (PCS 00-07-d9-00-45-7a)	

Name	PCS 00-07-d9-00-45-7a
Description	Proactive Communication Station
Location	
Password	•••••••

Viewing the IP address

To determine the IP address of a PCS 570/560 select the Favourites button ♥, then Settings and then Phone Information (alternatively press the key at the bottom right hand side and then select Phone Information). The IP address will be displayed at the top of the screen.

To determine the IP address of the PCS 580 press the Settings icon ⚙ five times. The IP address will be displayed at the top of the Phone Information screen.

A list of the IP addresses for each PCS 570/560/580 can be viewed in Manager by selecting Utilities and then IP Addresses.

31 users/devices have registered to this Call Server since last reboot.

IP Address	Port	Name found for address	Hardware Id
192.168.0.1		Modules.Craddock Call Server.BRI1	SpliceGateway 00-07-d9-00-18-23
192.168.0.247	5001	Users.Extn2012	SplicePOTS15 00-07-d9-00-04-95
192.168.0.248	5001	Users.Amy Stewart	SplicePhone 00-07-d9-00-18-f7
192.168.0.249	5001	Users.Sebastian Falks	SplicePhone 00-07-d9-00-15-43

13:06:04 Tuesday, 4th April 2006

Setting the IP Address

When connecting a PCS 560/570/580 to the system for the first time a DHCP server is required to give the PCS an IP address. This address can be viewed as described above. However once connected the PCS 560/570/580 can be given a static address as follows:

1. Within the Command prompt enter

   telnet <ip address of PCS 570/560>   eg telnet 192.168.0.250

2. You will be prompted for a Username, enter *diag*
3. You will be prompted for the password for this phone. By default this is *7388*. (Please refer to the Setting the Password section from page 53 for further details.)
4. Wait for a # to appear to confirm a connection has been made to the PCS
5. Enter *setnet*

6   The current configuration will be displayed. You will be asked if you wish to change any of this configuration.

7   Enter *y*

8   You will be asked how the IP address will be assigned.

9   Delete the *d* and enter *s*

10   Follow the onscreen prompts entering the IP address you wish to give the PCS, eg 192.168.0.50, the netmask you wish to use, eg 255.255.255.0, the IP address of the Gateway (Default Router), eg 192.168.0.1 and the IP address of the TFTP server which will be the IP address of the Call Server to provide software to the PCS during an upgrade, eg 192.168.0.1.

11   At any other prompt make the changes you require otherwise press Enter to leave the option unchanged.

12   You will be asked if you wish to save the changes. Enter *y*.

13   Telnet will display that the changes are being saved.

14   When finished enter *poweroff* and the PCS will reboot

Please note:
- If a PCS 560/570/580 has been given an IP address already being used on the network a warning message will be displayed on the screen.
- To return to using DHCP follow the same instructions above. Enter *d* when asked how the IP address will be assigned.
- At any prompt you may enter a *?* for help on that option or *q* to quit.
- For further information on working with Telnet please refer to page 270.
- Setnet can be used to set the Time Zone to be used by a specific IP phone

Connect to the Call Server remotely

If the PCS 580/570/560 is to be installed at a remote location, eg via a VPN connection, the phone should be configured with the IP address of the Call Server it is to connect to as follows.

1   Follow steps 1-7 as described above to telnet on to the phone and access Setnet.

2   Press Enter until the line "Startup arguments" appears.

3   Type in *–gk <ip address of call server>* eg –gk 192.168.0.1, and press Enter

4   Continue to press Enter until you are asked if you wish to save the changes. Enter *y*.

5   Telnet will display that the changes have been saved.

6   When finished enter *poweroff* and the PCS will reboot.

If the Call Server is to collect BLF data from this remote phone enter *–blf* after the IP address of the call server, eg –gk 192.168.0.1 –blf. The Call Server will then multicast this BLF information to all PCSs on the system.

If there are more than one remote IP Phone connected to the system one of these phones can be configured to collect the BLF data from all the remote phones and send this to the system. This can be achieved by entering *–relay* after the IP address, eg –gk 192.168.0.1 –relay. The other phones would then just be configured with the IP address of the call server, eg –gk 192.168.0.1.

It is recommended that no more than 5-6 remote IP phones should be configured to use this facility in order to keep the overhead on the Call Server to a minimum.
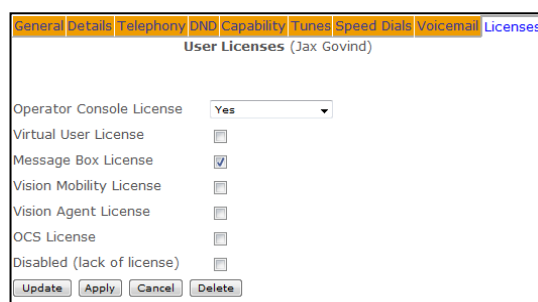
To remove this entry repeat steps 1 and 2 above. Delete the entry within Startup arguments and press Enter. Continue with steps 4, 5 and 6 above.

Operator Console Mode

When a PCS 580/570/560 is being used by a receptionist or switchboard operator the Operator Console Mode can be enabled to provide the operator with additional data to assist with the efficient handling of calls. The additional data is displayed within the User's Favourites screen and can display a User's forwarding settings, DND settings, when the User is logged out, when the User is on a call and who they are talking to, when a User is listening to voicemail, the number of new voicemail messages awaiting a User/Department and the number of missed calls received by a User/Department. Please refer to the relevant user manual for further details.

The Operator Console Mode is configured as follows:

1   Install an Operator Console licence as described from page 35.
2   Open the User configuration form for the User who wishes to use Operator Console Mode.
3   Select the Licences page
4   From the Operator Console Licence list box select either:
    a   Yes - to apply the licence to this User and enable Operator Console Mode, or
    b   PCS Partner only – this option can be used when a PCS 60 is partnering a PCS 5xx IP phone. By selecting this option the Operator Console Mode information will only be displayed on the PCS 60. This will prevent the IP phone from running out of memory when the User has configured many speed dials on a very active system.
5   Select Update or Apply when ready

PCS 410/400/100

Power to a PCS 410/400/100 is supplied by Power over Ethernet (PoE). This can be provided either by a third party PoE switch or directly via a LAN port available on a Call Server or 5330 Phone Module. Alternatively, the individual STEPS Power over Ethernet PSU can be used.
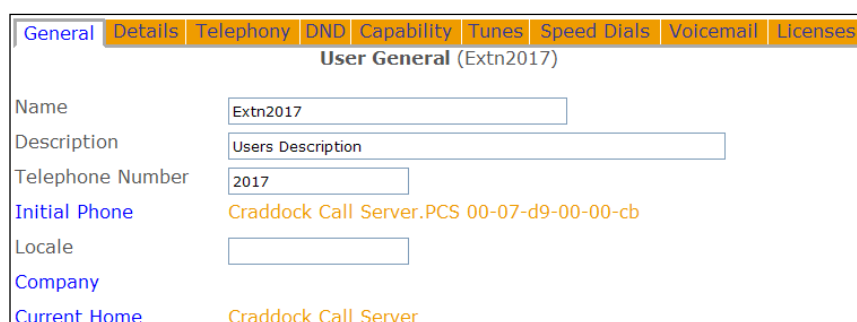
The PCS 400 can also be powered by a standalone PCS PSU connected directly to the back of the phone. When using these individual power supplies, it is recommended that when powering on/off the PCS 400 that this is done via the power block not via the connection to the phone. Please note: if a PCMCIA card is required in the PCS 400 a PCS PSU must be used.

Installing a PCS 410/400/100

1   Connect the PCS to the system via the network or directly to a LAN port on the Call Server or 5300 Phone Module
2   The PCS is a DHCP client and must obtain its IP address from a DHCP server.
3   An IPUser licence is required and must be installed before registration (See page 35 for further information.)
4   If the Auto Add Phones feature is turned on the Call Server will automatically register the PCS. (Please refer to page 34 for further details.)

If the Auto Add Phones feature is turned off:

a   In Manager select Unassigned Phones
b   The MAC address of the PCS will identify the unit connected. Click on the unit required
c   From the Status list box select Member
d   Select Update when ready
e   The unit will now appear under Phones

5   Once the system has registered the PCS a User will be automatically created, configured with the next available extension number.
6   The User's Initial Phone field displays the name of the PCS that is assigned to this User.

General	Details	Telephony	DND	Capability	Tunes	Speed Dials	Voicemail	Licenses

User General (Extn2017)

Name	Extn2017
Description	Users Description
Telephone Number	2017
Initial Phone	Craddock Call Server.PCS 00-07-d9-00-00-cb
Locale	
Company	
Current Home	Craddock Call Server

The above procedures also apply to non-SpliceCom IP Phones. A list of recommended IP Phones is available from our Marketing department.

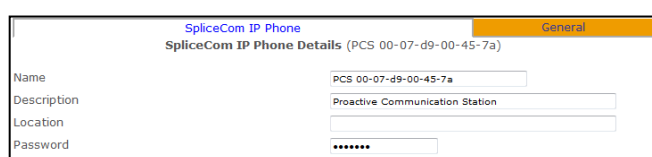Each IP Phone connected can be configured independently, see page 79 for further details.

Please note that a PCS 410/400 has no storage capabilities and therefore any downloads performed by the User will not be stored permanently.

Please refer to the PCS 400/410 and 50 User Manual and the PCS 100 User Manual for detailed descriptions of how to use these IP phones.

Setting the Password

When using Telnet to, for example, upgrade the phone or change its IP Address. The phone's password is entered to authorise these changes. By default this is set to 7388 but this password can be changed as follows:

1   In Manager, from the Navigation pane, select Phones
2   Select the module that the phone is connected to, eg Call Server
3   The MAC address of the unit will identify the specific phone required
4   Select the phone required
5   In the Password field enter the alpha-numeric characters required
6   Select Update or Apply to save the changes.

SpliceCom IP Phone	General

SpliceCom IP Phone Details (PCS 00-07-d9-00-45-7a)

Name	PCS 00-07-d9-00-45-7a
Description	Proactive Communication Station
Location	
Password	•••••••

Viewing the IP address

To determine the IP address of a PCS 410/400, on the handset press the More Options icon twice and then the Information icon.

The IP address of a PCS 100 is displayed during power up of the phone. Alternatively, after power up press the Down arrow once (or twice if Favourites are displayed) on the handset, the IP address will be displayed on the screen. Press the down arrow or up arrow to return to the previous screen.

A list of the IP Addresses for each PCS 410/400/100 can be viewed in Manager by selecting Utilities and then IP Addresses.

31 users/devices have registered to this Call Server since last reboot.

IP Address	Port	Name found for address	Hardware Id
192.168.0.1		Modules.Craddock Call Server.BRI1	SpliceGateway 00-07-d9-00-18-23
192.168.0.247	5001	Users.Extn2012	SplicePOTS15 00-07-d9-00-04-95
192.168.0.248	5001	Users.Amy Stewart	SplicePhone 00-07-d9-00-18-f7
192.168.0.249	5001	Users.Sebastian Falks	SplicePhone 00-07-d9-00-15-43

13:06:04 Tuesday, 4th April 2006

Setting the IP Address

When connecting a PCS 410/400/100 to the system for the first time a DHCP server is required to give the PCS an IP address. This address can be viewed as described above. However once connected the PCS 410/400/100 can be given a static address as follows:

1   Within the Command prompt enter

    telnet <ip address of PCS 410/400/100>   eg telnet 192.168.0.250

2   You will be prompted for a Username, enter *diag*
3   You will be prompted for the password for this IP phone. By default this is *7388*. (Please refer to the Setting the Password section from page 56 for further details.)
4   Wait for a # to appear to confirm a connection has been made to the PCS
5   Enter *setnet*
6   The current configuration will be displayed. You will be asked if you wish to change any of this configuration.
7   Enter *y*
8   You will be asked how the IP address will be assigned.
9   Delete the *d* and enter *s*
10 Follow the onscreen prompts entering the IP address you wish to give the PCS, eg 192.168.0.50, the netmask you wish to use, eg 255.255.255.0, the IP address of the Gateway (Default Router), eg 192.168.0.1 and the IP address of the TFTP server which will be the IP address of the Call Server to provide software to the PCS during an upgrade, eg 192.168.0.1.
11 At any other prompt make the changes you require otherwise press Enter to leave the option unchanged.
12 You will be asked if you wish to save the changes. Enter *y*.
13 Telnet will display that the changes are being saved.
14 When finished enter *poweroff* and the PCS will reboot

Please note:
- If a PCS 410, 400 or 100 has been given an IP address already being used on the network a warning message will be displayed on the screen.

- To return to using DHCP follow the same instructions above. Enter *d* when asked how the IP address will be assigned.
- At any prompt you may enter a *?* for help on that option or *q* to quit.
- For further information on working with Telnet please refer to page 270.
- Setnet can be used to set the Time Zone to be used by a specific IP phone
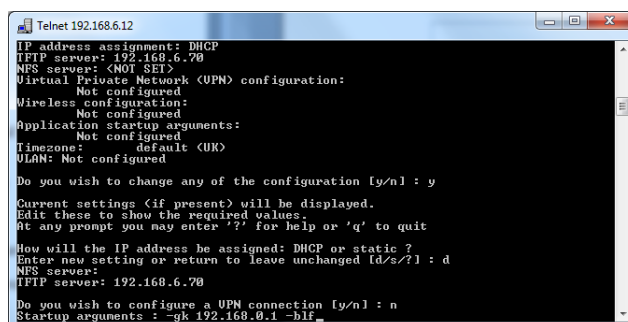
Connect to the Call Server remotely

If the PCS 410/400/100 is to be installed at a remote location, eg via a VPN connection, the phone should be configured with the IP address of the Call Server it is to connect to as follows.

1   Follow steps 1-7 as described above to telnet on to the phone and access Setnet.
2   Press Enter until the line "Startup arguments" appears.
3   Type in *–gk <ip address of call server>* eg –gk 192.168.0.1, and press Enter
4   Continue to press Enter until you are asked if you wish to save the changes. Enter *y*.
5   Telnet will display that the changes have been saved.
6   When finished enter *poweroff* and the PCS will reboot.

If the Call Server is to collect BLF data from this remote phone enter *–blf* after the IP address of the call server, eg –gk 192.168.0.1 –blf. The Call Server will then multicast this BLF information to all PCSs on the system.

If there are more than one remote IP Phone connected to the system one of these phones can be configured to collect the BLF data from all the remote phones and send this to the system. This can be achieved by entering *–relay* after the IP address, eg –gk 192.168.0.1 –relay. The other phones would then just be configured with the IP address of the call server, eg –gk 192.168.0.1.

It is recommended that no more than 5-6 remote IP phones should be configured to use this facility in order to keep the overhead on the Call Server to a minimum.

To remove this entry repeat steps 1 and 2 above. Delete the entry within Startup arguments and press Enter. Continue with steps 4, 5 and 6 above.

Installing PCS Software

PCS 60

The PCS 60 is a PC application that provides standard telephony functionality, CLI display, Operator Console mode and gives Users the ability to set features such as forwarding and DND via an easy to use dialogue box.   It can be run on MS Windows and Apple Mac OS X version 10.4 or above. It can be installed as an IP soft phone on a multi-media PC or programmed to partner a User's telephone handset (PCS 580, 570, 560, 410/400, 100, 10, 5 or an alternative analogue handset).

Please refer to the PCS 60 User Manual for detailed descriptions on how to use this software.

Installing the PCS 60 software on a Windows PC

1   The set up file will be supplied in a compressed folder called, for example, PCS60.3.2.31.msi.zip. Open this folder.
2   Run (double click) the set up file called, for example, PCS60.3.2.31.msi
3   At the Welcome screen select Next to continue the installation
4   By default the software will be loaded into C:\Program Files\SpliceCom\PCS60. Change this location if required.
5   Where relevant you will also have the option to load the software for all Users of the PC or just for the currently logged in User.
6   Select Next when ready
7   At the Confirm Installation screen select Next.
8   The software will be installed
9   At the Installation Complete screen select Close
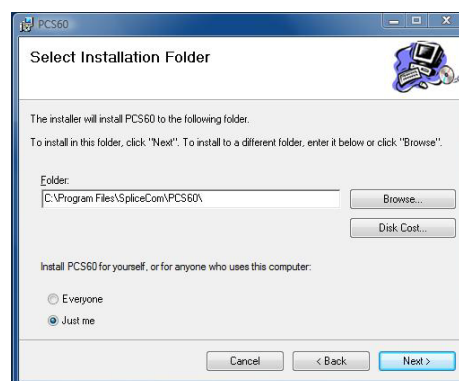
PCS 60 as an IP Soft Phone on a Windows PC

The PCS 60 software can be run as an IP soft phone where the PC becomes the User's telephone and the voice path will be via the PC. Therefore the PC will require a headset, microphone, sound card etc installed.

Before starting:
1   The PC must have a network connection to a Call Server.
2   An IPUser licence must be previously installed and be available. (For further details please refer to page 35.)

Running the PCS 60:
1   Run the PCS60.exe file either from the Installation folder, eg C:\Program Files\SpliceCom\PCS60, or create a short cut to this file in the required location and double click on the short cut.
2   The PCS 60 application will open.
3   The "Searching for Server" message will be displayed as the registration process on the Call Server is started.

   If the "Searching for Server" message remains and the PCS 60 does not continue to the next step check that the PC can communicate with the Call Server or refer to the Connect to the Call Server Remotely section below.

4   The "No Licences Available" message will be displayed while the phone is registered as described in step 5.
5   If the Auto Add Phones feature is turned on the Call Server will automatically register the PCS 60. (Please refer to page 34 for further details.)
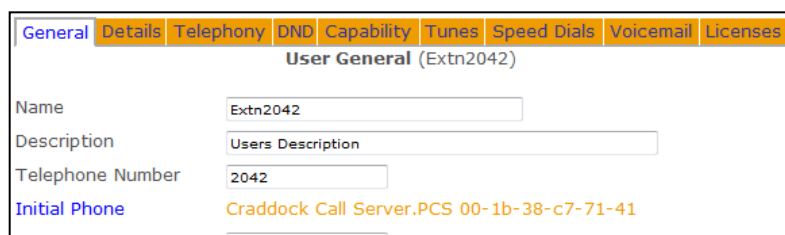
   If the Auto Add Phones feature is turned off:
a   In Manager select Unassigned Phones
b   The hardware address of the PC will identify the PCS 60 to be connected. Click on the entry required. (If the PCS 60 is not listed check the installation of the IPUser licence as detailed from page 35.)
c   From the Status list box select Member
d   Select Update when ready

e   The PCS 60 will now appear under Phones, identified by the hardware address of the PC

6   The "Security Access Denial" message will be displayed while an IPUser licence is assigned.
7   A User account will be automatically created configured with the next available extension number. This User will be assigned to the PCS 60, the "Security Access Denial" message will disappear and the PCS 60 will be ready for use. The name of the assigned User will be displayed in the Status Bar at the bottom left hand corner of the PCS 60.

The User's Initial Phone field displays the PCS 60 that the User is assigned to, identified by the hardware address of the PC.

Connect to the Call Server remotely

If the PCS 60 is to be run remotely over, a different subnet, a VPN, for example, the application must be configured with the IP address of the gatekeeper, in other words the Call Server, before the registration process can take place. This can be done in two ways:

1   Using PCS 60 Preferences

a   From the File menu select Preferences
b   In the IP Address field enter the IP address of the Call Server, eg 192.168.0.1
c   Select OK (Window only
d   Close down the PCS 60 and run the software again.

2   Using the PCS 60's short cut properties

a   Open the Properties of the PCS 60's short cut
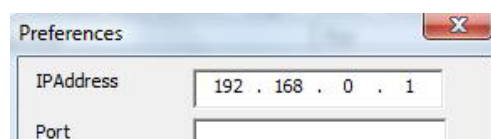
b   In the Target field add

-gk <ip address of Call Server>, eg –gk 192.168.0.1

so that the Target path reads as per the following example:

"C:\Program Files\SpliceCom\PCS60\PCS60.exe" –gk 192.168.0.1

c   Select OK.
d   Double click on the short cut to run the PCS 60.

Please note that entries made in the short cut Properties will override the entries made in the Preferences dialog box.

Remote BLF

If the call server is to collect BLF information from this remote PCS 60 IP Phone the Remote BLF option should be enabled as follows.

1   From the File Menu select Preferences
2   Tick Remote BLF
3   Select OK
4   Close down the PCS 60 and run the software again.

It is recommended that no more than 5-6 remote IP phones should be configured to use this facility in order to keep the overhead on the Call Server to a minimum.

PCS 60 on a Windows PC as a partner to a User's handset

The PCS 60 can be run as partner to any PCS or analogue handset being used on the **max**imiser system (PCS 520/505/10/5, alternative analogue handset, PCS 100, PCS 410/400, PCS 560, PCS 570 and PCS 580) giving the user the additional functionality provided by the PCS 60.

Before starting:

In order for the PCS 60 to partner a telephone the PCS 60 must connect to the port that the analogue handset is plugged into or to the relevant IP phone, therefore you will require the following information:
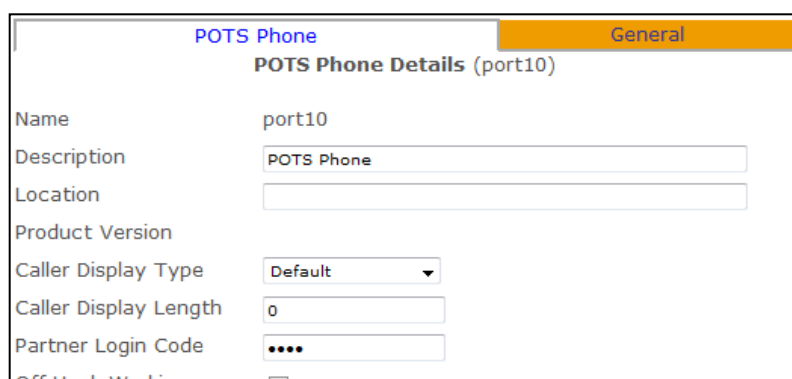
1   The IP address of the Phone Module or Call Server where the analogue port is located. This can be viewed within Manager. Select Modules and then select the relevant module, the IP address is displayed, eg 192.168.0.249.
     or
     The IP address of the PCS 560 or 570. This can be viewed by selecting the Favourites button, then Settings and then Phone Information, eg 192.168.0.226
     or
     The IP address of the PCS 580. This can be viewed within the Phone Information screen, which can be accessed by pressing the Settings icon 5 times.
     or
     The IP address of a PCS 410/400. This can be viewed by selecting the Information icon from the Home Page toolbar, eg 192.168.0.221
     or

The IP address of a PCS 100. This can be viewed by pressing the Down arrow once (or twice if Favourites are displayed) on the handset; the IP address being used by the handset will be displayed, eg 192.168.0.245. The IP address is also displayed during power up.

2   The reference number of the port:-
    a   Each analogue port on a Phone module or Call Server is given a reference number starting at 5001. Therefore port 1 is referenced as 5001, port 2 as 5002 etc.
    b   The port reference number for a PCS 580/570/560/410/400/100 will always be 5001.

3   The User's Partner Login Code. This can be configured as follows:

Configure a Phone's Partner Login Code
1   In Manager select Users
2   Select the User who wishes to use the PCS 60 as a partner to their handset
3   Click on the entry within the Initial Phone field eg Phone Module 1.port01
4   You will be taken to the port assigned to this User
5   In the Partner Login Code field enter a 1-8 digit access code, eg 1234

6   Select Update or Apply when ready.

Run the PCS 60 as a partner
1   Run the PCS60.exe file either from the Installation folder, eg C:\Program Files\SpliceCom\PCS60, or create a short cut to this file in the required location and double click on the short cut. (Please note that if you do not wish the PCS 60 to be loaded initially as an IP soft phone disconnect the PC from the network or turn off Auto Add Phones.)
2   The PCS 60 application will open.
3   From the File menu select Preferences
4   In the IP Address field enter the IP address of the Call Server or Phone Module or IP Phone as described above, eg 192.168.0.249
5   In the Port field enter the port number that the analogue phone is connected to, as described above, or 5001 for an IP phone.
6   In the Partner Login Code field enter the phone's Partner Login Code configured above.
7   Select OK when ready
8   Close down the PCS 60.
9   At this point your PC will need a network connection to the **max**imiser system.
10   Run the PCS 60 again.

11   Once a connection has been made the name of User assigned to the port that the PCS 60 is partnering is displayed in the Status Bar at the bottom left hand corner.
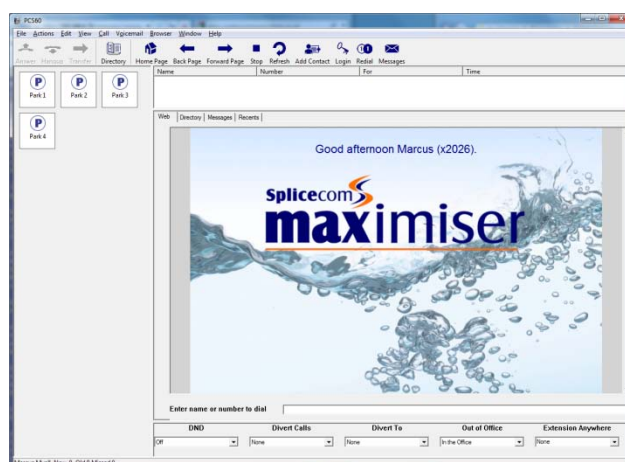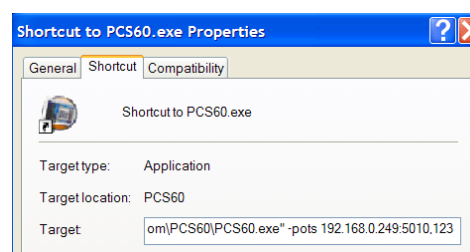

12   Lift the handset on the telephone to confirm that the connection between the PCS 60 and the phone is set up correctly. A call entry will be displayed in Call Status.

Alternatively, the IP address, port number and partner login code can be entered in the Properties of a short cut to the PCS 60 software. This method should be used when you wish to run more than one version of the PCS 60 on the same PC as described below on page 63.

(Please note that any entry made in the short cut's Properties will override entries in Preferences.)

1   Create a short cut to PCS60.exe.
2   Open the Properties of the short cut and in the Target field add the following:

-pots <ip address of Phone module>:<port no>,<partner login code>
or
 -pots <ip address of Call Server>:<port no>,<partner login code>
or
-pots <ip address of PCS 580/570/560/410/400/100>:<5001>,<partner login code>

so that the Target path reads as per the following example:

"C:\Program Files\SpliceCom\PCS60.exe" -pots 192.168.0.249:5010,1234

3   Select OK when ready
4   Double click on the short cut to run the PCS 60 application

Run more than one version of PCS 60 on the same MS Windows PC
When a member of staff is responsible for monitoring or using multiple handsets for example, multiple PCS 60 windows can be run on the same PC by selecting the Multiple Instances options within Preferences.

1   From the File menu select Preferences
2   Tick the Multiple Instances field
3   Select OK.

Create a short cut to the PCS 60 software for each PCS 60 window required. Configure the short cut properties for each connection required.
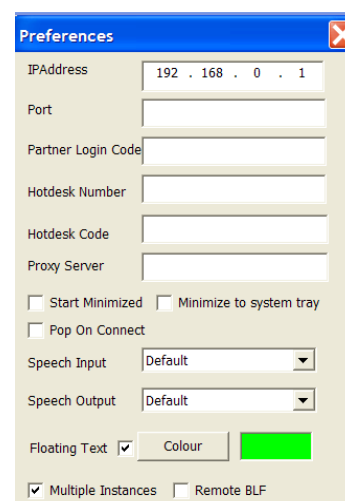
Installing the PCS 60 software on a Apple Mac OS X PC

The PCS 60 software will run on Mac OS X version 10.4 or above.

1   The set up file will be supplied in a compressed folder called, for example, PCS60_MAC.3.2.67.pkg.zip. Open this folder.
2   Run (double click) the set up file called, for example, PCS60_MAC.3.2.67.pkg
3   At the Welcome screen select Continue to start the installation
4   At the Select a Destination screen select the Volume where you want the software loaded
5   By default the software will be loaded into the Applications folder. Change this location if required.
6   Select Continue when ready
7   At the Standard Install screen select Install. (If you wish at this point to change the Install Location select Change Install Location.)
8   You will be prompted for your PC log in password, enter this in the Password field.
9   The software will be installed
10 At the Installation Completed screen select Close

PCS 60 as an IP Soft Phone on an Apple Mac PC

The PCS 60 software can be run as an IP soft phone where the PC becomes the User's telephone and the voice path will be via the PC. Therefore the PC will require a headset, microphone, sound card etc installed.

Before starting:
1   The PC must have a network connection to a Call Server.
2   An IPUser licence must be previously installed and be available. (For further details please refer to page 35.)
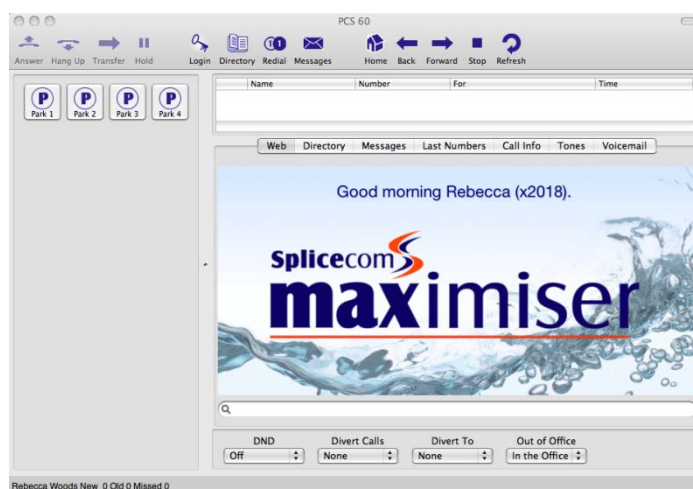
Running the PCS 60:
1   Run the PCS60 software either from Applications or create an Alias to this file in the required location and double click on the Alias.
2   The PCS 60 application will open.
3   The "Searching for Server" message will be displayed as the registration process on the Call Server is started.

If the "Searching for Server" message remains and the PCS 60 does not continue to the next step check that the PC can communicate with the Call Server or refer to the Connect to the Call Server Remotely section below.

4   The "No Licenses Available" message will be displayed while the phone is registered as described in step 5.
5   If the Auto Add Phones feature is turned on the Call Server will automatically register the PCS 60. (Please refer to page 34 for further details.)

If the Auto Add Phones feature is turned off:
a   In Manager select Unassigned Phones
b   The hardware address of the PC will identify the PCS 60 to be connected. Click on the entry required. (If the PCS 60 is not listed check the installation of the IPUser licence as detailed from page 35.)
c   From the Status list box select Member
d   Select Update when ready
e   The PCS 60 will now appear under Phones, identified by the hardware address of the PC

6   The "Security Access Denial" message will be displayed while an IPUser licence is assigned
7   A User account will be automatically created configured with the next available extension number. This User will be assigned to the PCS 60, the "Security Access Denial" message will disappear and the PCS 60 will be ready for use. The name of the assigned User will be displayed in the Status Bar at the bottom left hand corner of the PCS 60.


The User's Initial Phone field displays the PCS 60 that the User is assigned to, identified by the hardware address of the PC.

Connect to the Call Server remotely

If the PCS 60 is to be run remotely over, a different subnet, a VPN, for example, the application must be configured with the IP address of the gatekeeper, in other words the Call Server, before the registration process can take place.

1   From the PCS60 menu select Preferences
2   In the IP Address field enter the IP address of the Call Server, eg 192.168.0.1
3   Tab to the next field
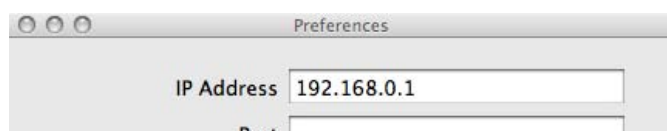
4   Close the window
5   Close down the PCS 60 and run the software again.

Remote BLF

If the call server is to collect BLF information from this remote PCS 60 IP Phone the Remote BLF option should be enabled as follows.

1   From the PCS 60 Menu select Preferences
2   Tick Remote BLF
3   Close down the PCS 60 and run the software again.

It is recommended that no more than 5-6 remote IP phones should be configured to use this facility in order to keep the overhead on the Call Server to a minimum.

PCS 60 on an Apple Mac PC as a partner to a User's handset

The PCS 60 can be run as partner to any PCS or analogue handset being used on the **max**imiser system (PCS 520/505/10/5, alternative analogue handset, PCS 100, PCS 410/400, PCS 560, PCS 570 and PCS 580) giving the user the additional functionality provided by the PCS 60.

Before starting:

In order for the PCS 60 to partner a telephone the PCS 60 must connect to the port that the analogue handset is plugged into or to the relevant IP phone, therefore you will require the following information:
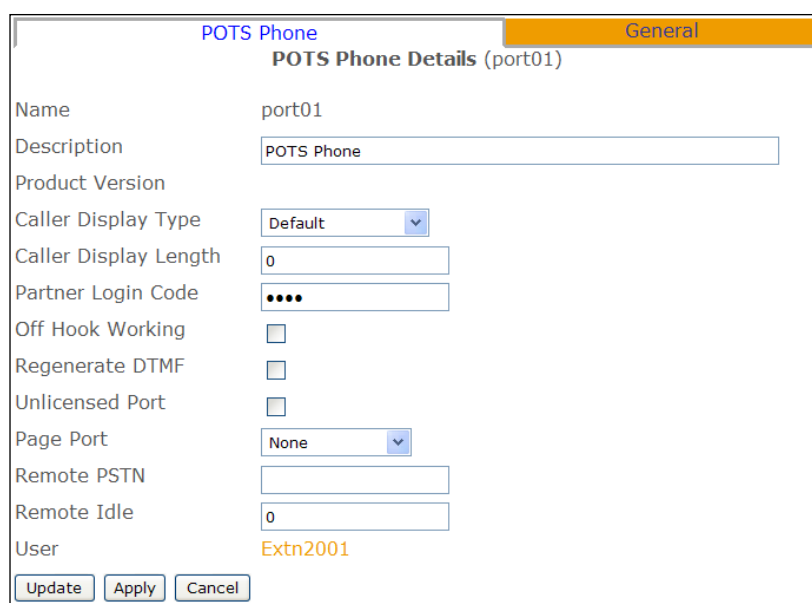
1   The IP address of the Phone Module or Call Server where the analogue port is located. This can be viewed within Manager. Select Modules and then select the relevant module, the IP address is displayed, eg 192.168.0.249.
     or
     The IP address of the PCS 560 or 570. This can be viewed by selecting the Favourites button, then Settings and then Phone Information, eg 192.168.0.226
     or
     The IP address of the PCS 580. This can be viewed within the Phone Information screen, which can be accessed by pressing the Settings icon 5 times.
     or
     The IP address of a PCS 410/400. This can be viewed by selecting the Information icon from the Home Page toolbar, eg 192.168.0.221
     or

The IP address of a PCS 100. This can be viewed by pressing the Down arrow once (or twice if Favourites are displayed) on the handset, the IP address being used by the handset will be displayed, eg 192.168.0.245. The IP address is also displayed during power up.

2   The reference number of the port:-
    a   Each analogue port on a Phone module or Call Server is given a reference number starting at 5001. Therefore port 1 is referenced as 5001, port 2 as 5002 etc.
    b   The port reference number for a PCS 580/570/560/410/400/100 will always be 5001.

3   The User's Partner Login Code. This can be configured as follows:

Configure a Phone's Partner Login Code
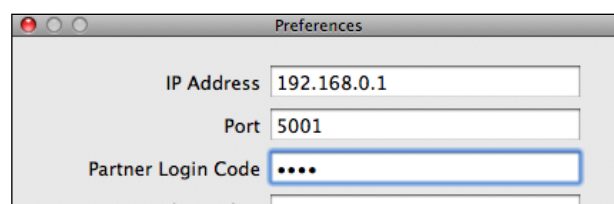1   In Manager select Users
2   Select the User who wishes to use the PCS 60 as a partner to their handset
3   Click on the entry within the Initial Phone field eg Phone Module 1.port01
4   You will be taken to the port assigned to this User
5   In the Partner Login Code field enter a 1-8 digit access code, eg 1234

POTS Phone	General
POTS Phone Details (port01)	

Name	port01
Description	POTS Phone
Product Version	
Caller Display Type	Default
Caller Display Length	0
Partner Login Code	••••
Off Hook Working	☐
Regenerate DTMF	☐
Unlicensed Port	☐
Page Port	None
Remote PSTN	
Remote Idle	0
User	Extn2001

Update   Apply   Cancel

6   Select Update or Apply when ready.

Run the PCS 60 as a partner
1   Run the PCS60 software either from Applications or create an Alias to this file in the required location and double click on the Alias. (Please note that if you do not wish the PCS 60 to be loaded initially as an IP soft phone disconnect the PC from the network or turn off Auto Add Phones.)
2   The PCS 60 application will open.
3   From the PCS60 menu select Preferences
4   In the IP Address field enter the IP address of the Call Server or Phone Module or IP Phone as described above, eg 192.168.0.249

Preferences	
IP Address	192.168.0.1
Port	5001
Partner Login Code	••••

5   In the Port field enter the port number that the analogue phone is connected to, as described above, or 5001 for an IP phone.

6   In the Partner Login Code field enter the phone's Partner Login Code configured above.Tab to the next field

7   Close the window when ready

8   Close down the PCS 60.

9   At this point your PC will need a network connection to the **max**imiser system.

10   Run the PCS 60 again.

11   Once a connection has been made the name of User assigned to the port that the PCS 60 is partnering is displayed in the Status Bar at the bottom left hand corner.
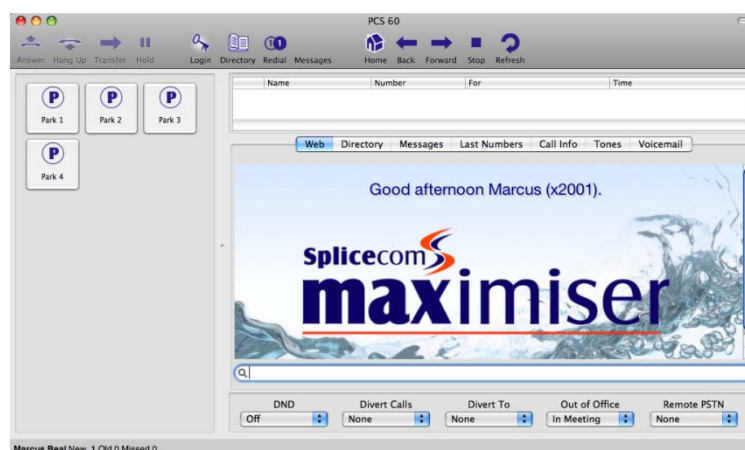
12   Lift the handset on the telephone to confirm that the connection between the PCS 60 and the phone is set up correctly. A call entry will be displayed in Call Status.

Run more than one version of PCS 60 on the same Apple Mac PC

When a member of staff is responsible for monitoring or using multiple handsets for example, multiple PCS 60 windows can be run on the same PC by making a copy of the software and entering the required settings in the Preferences form of each copy of the PCS 60 software. The Preference settings are saved with each copy of the software.

Operator Console Mode

When a PCS 60 is being used by a receptionist or switchboard operator the Operator Console Mode can be enabled to provide the operator with additional data to assist with the efficient handling of calls. The additional data is displayed within the User's Favourites screen and can display a User's forwarding settings, DND settings, when the User is logged out, when the User is on a call and who they are talking to, when a User is listening to voicemail, the number of new voicemail messages awaiting a User/Department and the number of missed calls received by a User/Department.
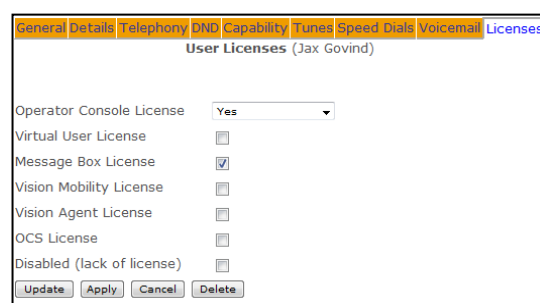
The operator can also be given an additional window to display further Speed Dials when there is insufficient room in the Favourites pane.

For further details please refer to the PCS 60 User Manual.

The Operator Console Mode is configured as follows:

1   Install an Operator Console licence as described from page 35.

2   Open the User configuration form for the User who wishes to use Operator Console Mode.

3   Select the Licences page

4 From the Operator Console Licence list box select either:

a Yes - to apply the licence to this User and enable Operator Console Mode, or

b PCS Partner only – this option can be used when a PCS 60 is partnering a PCS 5xx IP phone. By selecting this option the Operator Console Mode information will only be displayed on the PCS 60. This will prevent the IP phone from running out of memory when the User has configured many speed dials on a very active system.

5 Select Update or Apply when ready

Further configuration of the Operator Console Mode can be done within the Preferences dialogue box, please refer to the PCS 60 User Manual for further details.

Administrator Access

A User of a PCS 60 can also be given the ability to change a User's DND and Forwarding settings via their Favourites. To enable this facility the operator will require an Administrator name and password. Please refer to page 31 for information on how to create an Administrator. Please refer to the PCS 60 User Manual for information on how to configure this facility.

PCS 50

The PCS 50 application provides the same telephony functionality as available on the PCS 410/400. This application can run as an IP soft phone or be programmed to partner a User's handset from their PC. The PCS 50 application can be installed on a Windows PC or Mac OS X.

Please refer to the PCS 400/410 and 50 User Manual for detailed descriptions of how to use this software.

Installing the PCS 50 software on a Windows PC

1 Run the PCS 50 set-up file called, for example, PCS50.2.2.652.setup.exe
2 At the Setup screen select Yes to continue with the installation.
3 At the Welcome screen select Next to continue
4 Select where on the PC you wish the software to be loaded. By default, this will be C:\Program Files/PCS50. Select Next when ready.
5 Enter the name that you wish to use for the Start Menu folder that will contain the PCS 50 short cut. By default, this will be PCS50. Select Next when ready.
6 Select whether you wish a short cut to be created on your desktop. By default, this option will be ticked. Select Next when ready.
7 The Ready to Install screen will confirm your previous choices. Select Back to make any changes. Select Install when ready.
8 At the Completing the PCS Setup Wizard screen select Finish.

PCS 50 as an IP Soft Phone on a Windows PC

The PCS 50 software can be run as an IP soft phone where the PC becomes the User's telephone and the voice path will be via the PC. Therefore the PC will require a headset, microphone, sound card etc installed.

Before starting:

1 The PC must have a network connection to a Call Server.
2 An IPUser licence must be previously installed and be available. (For further details please refer to page 35.)

Running the PCS 50:

1 Run the PCS.exe file either from the Installation folder, eg C:\Program Files\SpliceCom\PCS50, or via the short cut created during the installation
2 The PCS 50 application will open.
3 The PCS 50 application will use the same method as the PCS 60 to register with the system as described from step 3 on page 59.

Please note that the PCS 50 obtains the time from the DHCP Server.

PCS 50 as a partner to a User's handset on a Windows PC

The PCS 50 can be run to partner any PCS or analogue handset being used on the **max**imiser system (PCS 520/505/10/5, alternative analogue handset, PCS 100, PCS 410/400 and PCS 580/570/560) giving the user the additional functionality provided by the PCS 50.

The PCS 50 software uses the same method as the PCS 60 to partner an analogue handset as described from page 61.

Please note that the PCS 50 software is stored as PCS.exe in the Installation folder, eg C:\Program Files\SpliceCom\PCS50, or can be access via the short cut created during the installation

Installing the PCS 50 software on a Mac OS X

1 The set up file will be supplied in a compressed folder called, for example, PCS50.2.2.667.pkg.zip. Open this folder.
2 Run (double click) the set up file called PCS50
3 At the Welcome screen select Continue
4 At the Select a Destination screen select the Volume where you want the software to be loaded
5 By default the software will be loaded into the Applications folder. Change this location if required.

6   Select Continue when ready
7   At the Standard Install screen select Install. (If you wish at this point to change the Install Location select Change Install Location.)
8   You will be prompted for your PC log in password, enter this in the Password field.
9   The software will be installed
10   At the Installation Completed screen select Close

PCS 50 as IP Soft phone on a Mac OS X

The PCS 50 software can be run as an IP soft phone where the PC becomes the User's telephone and the voice path will be via the PC. Therefore the PC will require a headset, microphone, sound card etc installed.

The PCS 50 will use the same method as the PCS 60 to register with the system as detailed from page 64.

PCS 50 as a Partner to a User's handset on a Mac OS X

The PCS 50 can be run to partner any PCS or analogue handset being used on the **max**imiser system (PCS 520/505/10/5, alternative analogue handset, PCS 100, PCS 410/400, PCS 580/570/560) giving the user the additional functionality provided by the PCS 50.

The PCS 50 will use the same method as the PCS 60 to partner a User's handset as detailed from page 66.

PCS Operators Console

The PCS Operators Console application provides the most frequently used features and facilities of the PCS 50 on a single page, full screen display. This application has been specifically developed for Operators & Receptionists and can run as an IP soft phone or be programmed to partner a User's handset from their PC. The PCS Operators Console application can be installed on PCs or laptops running the Microsoft Windows or Apple Mac OS X operating systems. Please refer to the PCS Operators Console manual for details on working with this application.

Configuration of the PCS Operators Console is identical to that for the PCS 50:

- To install the PCS Operators Console on a Windows PC please refer to page 69.
- To configure the PCS Operators Console as an IP soft phone on a Windows PC please refer to page 69.
- To configure the PCS Operators Console as a Phone Partner on a Windows PC please refer to page 70.

- To install the PCS Operators Console on a Apple Mac OS X platform please refer to page 70.
- To configure the PCS Operators Console as an IP soft phone on an Apple Mac OS X platform please refer to page 71.
- To configure the PCS Operators Console as a Phone Partner on an Apple Mac OS X platform please refer to page 71.

Please note that a valid Operator Console Licence must be purchased for each User wanting to use this application. Once the appropriate licences have been entered on the system, enable the Operator Console Licence tick-box for each User who will be using the application. This is located within the Licences page of the User's configuration form.

Please refer to the Installing Licences section from page 35 for further information.

Connecting Multiple Call Servers

Two or more Call Servers can be connected to create one system where one will act as the Primary Call Server and all Call Servers connected to the Primary Call Server become Secondary Call Servers. This can be via the LAN or via a WAN link.

On power up, by default, each Call Server will use 192.168.0.1 as its IP address and act as a DHCP Server. If this feature is not required because, for example, 192.168.0.1 is already being used and another DHCP Server exists on the network do not connect the new Call Server to the network until this default configuration has been changed as described on page 28.

Linking two Call Servers across a LAN

1   Ensure you are running the same version of software at both ends.

2   On the Primary Call Server:
    a   In Manager select Modules and select the Call Server from the list
    b   You may wish to give the Call Server a unique name to help identify this Call Server eg Primary Call Server
    c   Select the Call Server page and note the Administration Mode field is set to Primary

3   On the Secondary Call Server:
    a   Power up the Secondary Call Server
    b   This Call Server must have a unique IP address and be on the same network as the Primary Call Server. Therefore, if necessary, change the IP address of the Call Server as described on page 28. It is recommended that this address be outside the address range given out by the DHCP Server, eg 192.168.0.252
    c   You may wish to the rename the Call Server to help identify this Call Server, eg Secondary Call Server.
    d   Power down your Call Server, as described on page 255.
    e   Connect the Call Server to the Primary Call Server either via the network or directly to a LAN port on the Primary Call Server.
    f   Power up the Call Server
    g   Open Manager on the Secondary Call Server, eg http://192.16.0.252/manager
    h   Select Utilities
    i   Under the Commands list select Ask Master
    j   Within the Enter IP Address field enter the IP Address of the Primary Call Server, eg 192.168.0.1
    k   Click on the Set Master button

4   On the Primary Call Server:
 a   In Manager select Unassigned Modules
 b   Select the new Call Server, eg Secondary Call Server
 c   From the Module Status list box select Member
 d   Select Update or Apply when ready
 e   Select Modules, select the Secondary Call Server, select the Call Server page and note that the Administration Mode is set to Secondary.

The Secondary Call Server will now be a part of the system controlled by the Primary Call Server. The Primary Call Server will replicate the relevant configuration information to the Secondary Call Server. This means that the configuration of the system can be administered from either Call Server.

Linking two Call Servers across a WAN link

1   Ensure you are running the same version of software at both ends.

2   On the Primary Call Server:
 a   This Call Server must be on different network to Secondary Call Server. Therefore, if necessary, change the IP address of the Call Server as described on page 28, eg 194.168.0.1. If this Call Server is to act as the DHCP Server change the DHCP properties as described on page 29, eg
 b   Start Address = 194.168.0.2
 c   End Address = 194.168.0.250
 d   Base Address = 194.168.0.0
 e   In Manager, select Modules and select the Call Server from the list
 f   You may wish to give the Call Server a unique name to help identify this Call Server eg Primary Call Server
 g   Select the Call Server page and note the Administration Mode field is set to Primary

3   On the Secondary Call Server:
 a   Power up the Secondary Call Server
 b   This Call Server must be on different network to Primary Call Server. Therefore, if necessary, change the IP address of the Call Server as described on page 28, eg 192.168.1.1. If this Call Server is to act as the DHCP Server change the DHCP properties as described on page 29, eg
  Start Address = 192.168.1.2
  End Address = 192.168.1.250
  Base Address = 192.168.1.0
 c   In Manager, select Modules and select the Call Server from the list. You may wish to the rename the Call Server to help identify this Call Server, eg Secondary Call Server.
 d   In Manager select WAN Links and select the Secondary Call Server
 e   Select the default WANLink. (For further details please refer to page 247.)
 f   Select the IPRoutes page
 g   Remove all the existing routes and add an entry to route back to the Primary Call Server, eg
  IP Address = 194.168.0.0
  IP Mask = 255.255.255.0
  Metric = 1
 h   Power down your Call Server, as described on page 255.
 i   Connect the Call Server to the Primary Call Server via your WAN link.
 j   Power up the Call Server.
 k   The WAN LEDs on both Call Servers will be solid if a connection has been made. You may wish to ping the Primary Call Server to verify that you can communicate with the Primary, eg ping 194.168.0.1

l   In Manager select Utilities then Commands
m   Under the Commands list select Ask Master
n   Within the Enter IP Address field enter the IP Address of the Primary Call Server, eg 192.168.0.1
o   Click on the Set Master button

4   On the Primary Call Server:
a   In Manager select Unassigned Modules.
b   Select the new Call Server, eg Secondary Call Server
c   From the Module Status list box select Member.
d   Select Update or Apply when ready
e   Select Modules.
f   Select the Secondary Call Server, select the Call Server page and note that the Administration Mode is set to Secondary.

5   On the Secondary Call Server
a   In Manager select the WAN Links.
b   Select the Secondary Call Server and select WANLink
c   Select the IPRoutes page and note that the default routes have reappeared.
d   Repeat step 3(g) above.
e   You may wish to ping the Primary Call Server to verify that you can communicate with the Primary, eg ping 194.168.0.1

The Secondary Call Server will now be a part of the system controlled by the Primary Call Server. The Primary Call Server will replicate the relevant configuration information to the Secondary Call Server. This means that the configuration of the system can be administered from either Call Server.

Connecting a Phone or Trunk Module to the Secondary Call Server
A Phone or Trunk module can be connected to the Secondary Call Server as described on page 39 and page 43.

The Call Server field within the Phone or Trunk Module's configuration form will display the name of the Call Server that the module it is connected to, ie the name of the Secondary Call Server.

When connecting a Phone Module to a Secondary Call Server the additional Users created are all listed under the Users list and have taken the next available extension numbers. Their Initial Phone field displays which port on which Phone Module they are using and their Current Home field, which determines which Call Server is providing their voicemail service, displays the name of the Secondary Call Server.

Connecting a PCS to the Secondary Call Server
A PCS 580/570/560, PCS410/400, PCS 100, PCS 60 IP phone or PCS 50 IP phone can be connected to the Secondary Call Server as described in the Connecting a PCS section from page 48.

The Initial Phone field within the User's configuration form will display which Call Server the PCS is connected to, ie the name of the Secondary Call Server. Please refer to page 81 for further details.

Select Phones and then the relevant Call Server to display which PCSs are connected to this Call Server.

Connecting an existing Primary Call Server to another Primary Call Server
Where a Call Server has been previously working as a standalone system and is now to be connected to another system the same method described above is used however the following should be noted:

- Any configuration (other than the IP address and Call Server Name) previously stored on the Secondary Call Server will be lost as this Call Server becomes a part of the Primary Call Server's system and takes its configuration from the Primary Call Server.
- Any Phone and/or Trunk modules and/or PCS IP phone should be disconnected from the Secondary Call Server before it is connected to the Primary Call Server. They can be reconnected again as described above.

A Global System

After the above procedures have taken place each Call Server, Phone Module, Trunk Module, PCS 580/570/560, 410/400, 100, PCS 60 IP phone and PCS 50 IP phone is part of the one system regardless of their location. The Primary Call Server will replicate the relevant configuration information to each Secondary Call Server.

Primary Call Server

Secondary Call Server
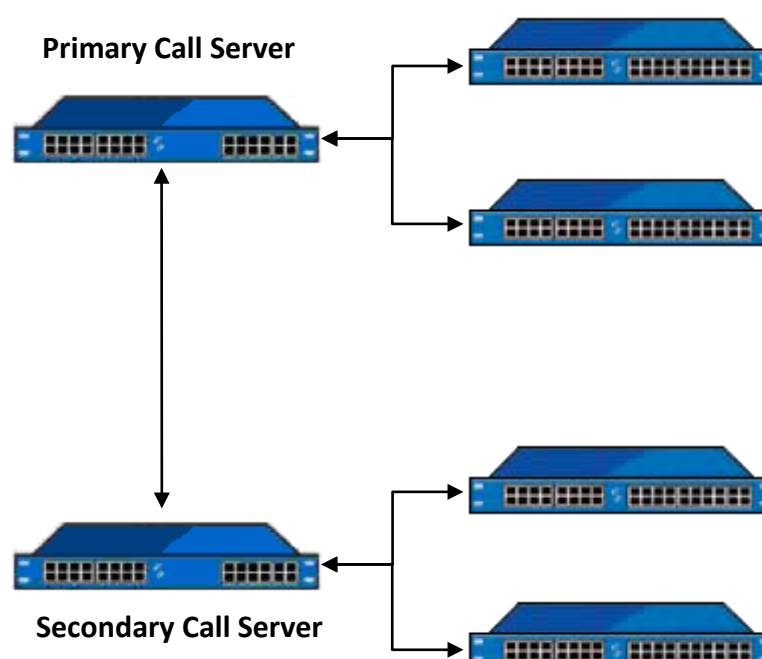
Each Secondary Call Server will be responsible for passing on relevant configuration information to Phone and Trunk Modules, IP Phones etc that are directly connected to this Call Server. Any configuration information received from these Phone Modules, Trunk Modules or IP Phones etc will be passed back to the Primary Call Server by the Secondary Call Server for storage in the master database.

If the Primary Call Server fails all Secondary Call Servers will continue to operate and devices that were registered to the Primary Call Server then re-register to a Secondary Call Server (provided that they are on the same LAN). This will be the first Secondary Call Server to respond.

If a device, eg Phone Module or PCS IP phone, is connected to the system via the network the Primary Call Server will always respond the quickest to a registration request, followed by the Secondary Call Server with the smallest number of devices attached and so on. This will ensure that the load is shared among all Secondary Call Servers. Therefore, to ensure that any module or IP Phone connected to the Secondary Call Server does not register with the Primary Call Server before the linking process is complete power up the unit after the Secondary Call Server has connected to the Primary Call Server.

User Status

To verify user registration across multiple Call Servers from Manager select Utilities and then User Status. This facility will display which Users are registered to which Call Server. All instructions are detailed within this page.
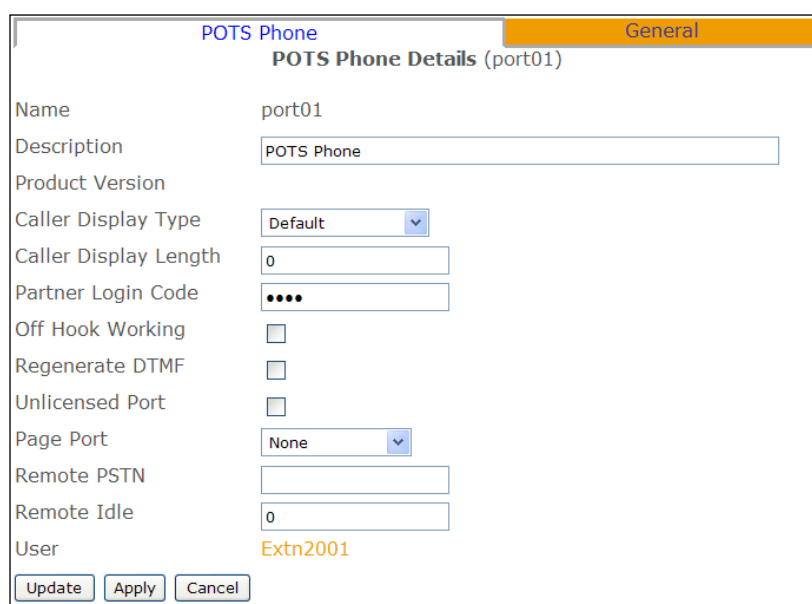
System Time

Each Call Server within one system can have a different date, time and Time Zone. Please refer to page 30 for further information.

Configuring PBX functionality

Configuring an Analogue Extension Port

Physical analogue extension ports on a Phone module or Call Server can be configured as follows:

1   In Manager select Phones
2   Select the relevant Phone Module or Call Server
3   Select the port you wish to configure

The port is automatically assigned a name which cannot be changed. This name is used by a User's Initial Phone field to indicate the port to which they are assigned. The User field displays the name of the User assigned to this port. Click on the name within the User field to view this User. Use the Cancel button to return to the port.

You may wish to enter a description to assist with identifying the port eg its physical location etc.

For a list of which ports have been assigned to which Users select Utilities and then Sorted Lists. This feature will also highlight a record where a port has been assigned to more than one User in error.

The following options can also be configured:

Caller Display Type
None – Call Display information will not be sent to the extension and the extension will not ring once after the relevant Phone Module has been rebooted
Default – PCS 10 (suitable for most analogue handsets)
Alarm – for use with alarm systems, DTMF is not sent out but allows the alarm system to send DTMF tones back out.
Bell – USA
BT – UK
BTNoMessages – for use with InterQuartz Discovery analogue phones
DTMF – Europe

Fax – for use with a fax machine connected to this port, provides a strong/robust ring tone improving the fax machine's ability to answer a call

FaxServer – for Fax Server support, please refer to page 182 for further details.

Lamp – for use with a PCS 5 to support the message waiting lamp

Interquartz – for use with InterQuartz Gemini analogue phones

Pay Phone – for use with a pay phone

PCS 10 – for use with the PCS 10

PCS 10 Lamp – for use with a PCS 10 to support the message waiting lamp

System Phone – for use with the PCS 520, ensures CLI information is displayed correctly and enables paging and auto answer functionality

Trunk – Sends a disconnect signal to the end device.

Caller Display Length

By default the Caller Display Length is set to 0, which means 20 characters will be sent to the handset. Change this length when more or less characters are required. Set to 16 for the PCS 10.

The information sent will display the Caller information and then the Destination information, eg John Barrow – Support. The destination uses the first word of, eg the User name, and remaining characters are used for the Caller's information.

Offhook Working

Tick this box if a headset is used in conjunction with the phone connected to this port. This feature allows the use of a headset without picking up the receiver.

Page Port

This field is used to determine the functionality of this analogue port. Select one of the following three options:

None – this port will be used as a standard extension with an analogue phone.

Paging – this port will be used by an external paging source, for example, a tannoy or PA system. A Paging Port Adapter is required to connect the powered speakers (please refer to page 340 for further details). This port can be paged via a Dial Plan entry, an example of which is available on page 120. Alternatively, voicemail can be used play a message over this port as described on page 209.
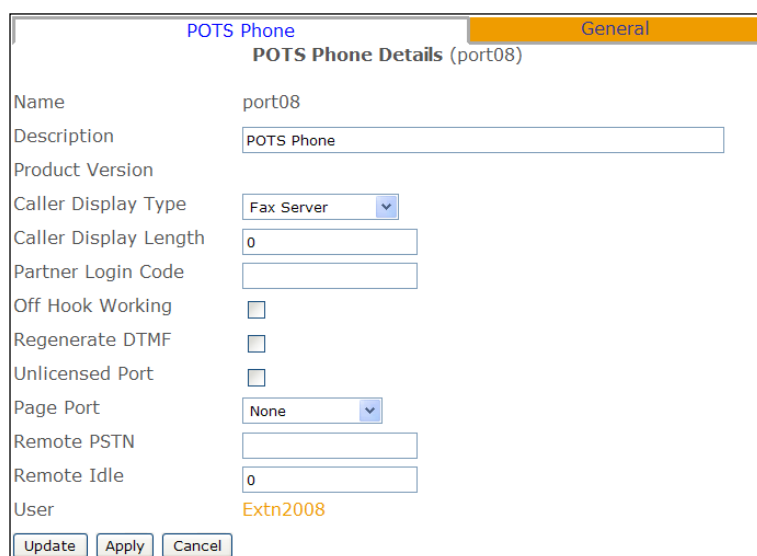
MusicOnHold – this port will be used to provide Music On Hold from an external source. Please refer to page 177 for further details.

Remote PSTN

For information on the Remote PSTN and Remote Idle fields please refer to the Setting Up Remote PSTN Operation section from page 111.
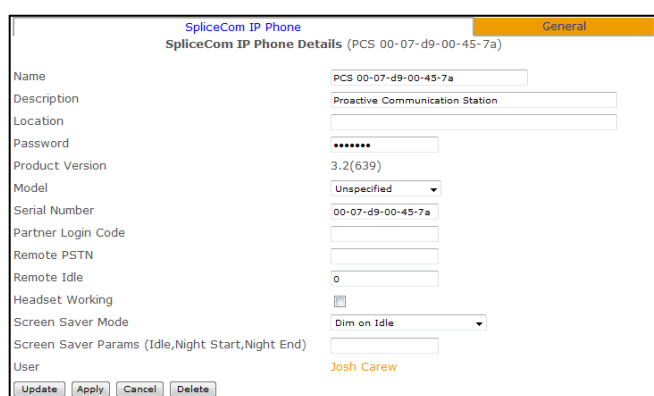
Fax Server Support

1   Configure the Caller Display Type on the analogue extension port connected to the Fax Server to "FaxServer".

2   Create a new Group and add the User assigned to the above port. (Please refer to page 122 for information on creating a Group.)

3   Create a new Department and in the Distribution Group add the Group created above. (Please refer to page 128 for information on creating a Department.)

4   Create a DDI Call Plan entry routed to this Department. (Please refer to page 169 for information on creating a DDI Call Plan.)

Configuring an IP Phone

1   In Manager select Phones

2   Select the module that the PCS 60 IP Phone, PCS 50 IP Phone, PCS 100, 400, 410, 560, 570, 580 or IP phone is connected to, eg Call Server

3   The MAC Address of the unit will identify the specific phone or PC

4   Select this entry if configuration is required.

You may wish to change the Name and/or the Description to assist with identifying the phone eg its physical location etc. This name is used by a User's Initial Phone field to indicate the IP phone to which they are assigned. The User field displays the name of the User assigned to this port. Click on the name within the User field to view this User. Use the Cancel button to return to the IP Phone.

For a list of which IP Phones have been assigned to which Users select Utilities and then Sorted Lists. This feature will also highlight a record where an IP Phone has been assigned to more than one User in error.

The following options can also be configured. Please note that the PCS 410/400 must be rebooted to activate any change of configuration.

Model
This field should be set to 'Colour640-480' for a PCS 410/400; otherwise this field should remain in default (Unspecified).

Screen Saver Mode
This feature can be used with the PCS 580, 570, 560, 410 and 400 and is used to enable the appearance of a screen saver. This can either be after the handset has been idle for a specified amount of time or during a specified time. In all cases the screen will return to normal by making the handset active, eg on a PCS 570 by picking up the handset, on a PCS 580 by touching the screen.

1   In the Screen Saver Mode field select one of the following options:

Blank on Idle	The screen saver will appear after the handset has been idle for the number of seconds indicated in the Screen Saver Params field.
Blank at Night	The Screen Saver will appear between the times entered in the Screen Saver Params field
Bright always	The screen saver will not be used.

2   In the Screen Save Params field enter the parameters required as *Idle Timeout (Secs),Night Start Time,Night End Time* eg 900,0,0 or 0,1730,0800 as follows:
   a   If the Screen Save Mode field is set to Blank on Idle enter the number of seconds that the handset is to remain idle before the screen saver will be appear, eg 900,0,0
   b   If the Screen Save Mode field is set to Blank at Night enter the time in 24 hour clock that the screen saver will appear followed by the time in 24 hour clock that the screen saver will be removed eg 0,1730,0800

3   Select Update or Apply when ready

Working with Users

The Users option within Manager displays each user of the **max**imiser system. These can be Users assigned to a physical analogue extension or IP Phone, or "floating" users who are not assigned to a specific location and will log onto a phone as and when required, or virtual users who do not need to use a physical phone but require the use of a feature of the system. PBX functionality is assigned to Users rather than to extension ports so that users of the system can log on to any phone and their User settings will follow them.
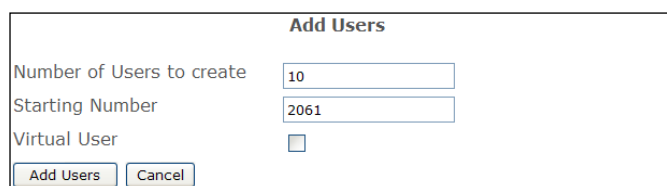
Creating a new User

When a 5100 or 5108 Call Server, 4140 Remote Call Server or Phone Module is connected to the system 4, 8, 15, 16 or 30 Users are automatically created and named, for example, Extn2017, Extn 2018 etc. Each User is assigned to an extension port, for example, Port01, Port02 etc, and given an extension number within the next available extension range, for example, 2017, 2018 etc.

When a PCS 60 (IP Phone), PCS 50 (IP Phone), PCS 100, PCS 410/400 or PCS 570/560 is installed a new User is automatically created and named, for example, Extn2031 and given the next available extension number, for example, 2031.

New Users can also be created manually, as follows, when a User is required but does not need to be assigned to a physical phone.

1   In Manager select Users
2   Select the Add button
3   In the Number of Users to Create field enter the number of Users you wish to create.
4   In the Starting Number field enter the extension number you wish to give the first User and any remaining Users will be given the next number within the range. For example, if you wish to create 10 Users with an extension range of 2061-2070 enter 2061 in this field.
5   Un-tick the Virtual User field unless you wish these Users to be virtual Users and the relevant licences have been installed (please refer to page 96 for further details on virtual Users).
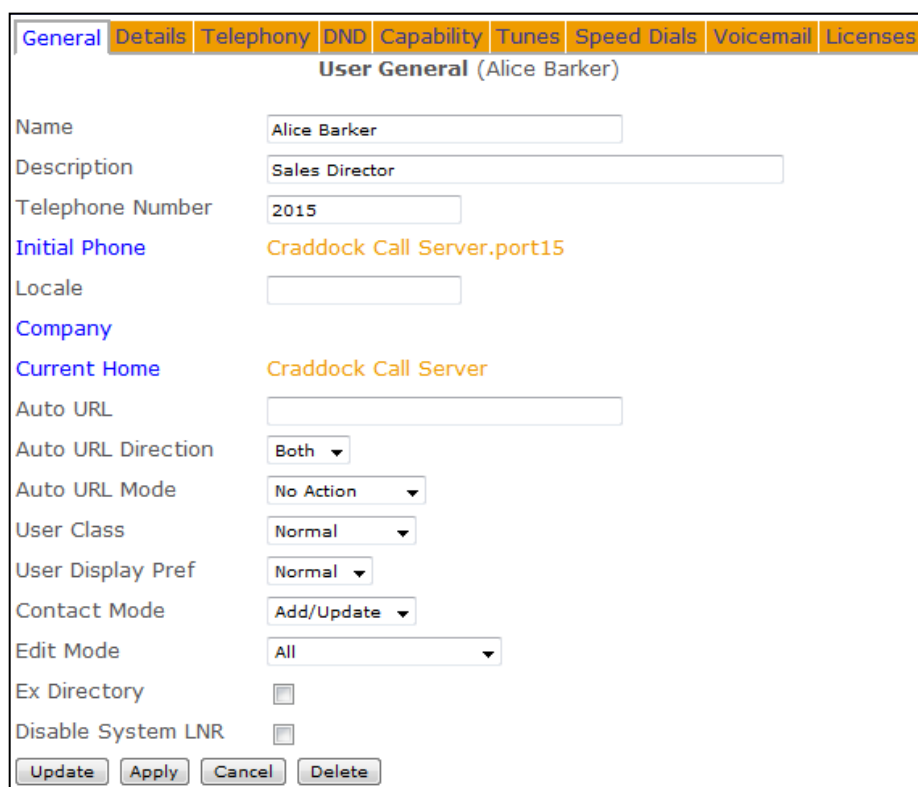
Add Users	
Number of Users to create	10
Starting Number	2061
Virtual User	☐
Add Users Cancel	

6   Select Add Users
7   You will be returned to the Users List and the new Users created will be listed at the top and named, for example, Extn2061, Extn2062 etc.
8   Each User can be selected and configured as describe below.

Configuring a User

1   In Manager select Users
2   Click on the relevant User name to view the configuration of each specific User.

Each User is assigned a Name which is used to identify them on the system, for example, when calls to and from this User are made the name will appear in Caller Display. It is recommended that this be changed to a unique, relevant name. The name should consist of alphanumerical characters only, not contain any punctuation and begin with a letter.

Each User is identified on the database by a GUID (Global Unique Identifier). This unique identifier can be displayed by selecting Utilities and then Voicemail GUIDs.

The Description field is a text field which can be used to further assist with identification of the User in Manager and via the Users directory accessed on the PCS 580/570/560/410/400/60/50, eg Personnel Director, Security, Reception etc

The Telephone Number field determines the internal number of the User's telephone.

The Initial Phone field indicates the physical analogue port or IP phone that is assigned to this User. Leave this field blank if the User is to log on to an available phone (please refer to the Hot Desking section on page 93 for further information).

The User's Locale specifies the language to be displayed on their IP phones. If left blank the System Locale will be used.

The Company field determines the Company to which this User is assigned. Please refer to the Working with Companies section from page 155 for further information.

The Current Home field determines which Call Server is providing the User with a voicemail service. Please refer to the Working with Voicemail section from page 187 for further information.

The User Class list box determines the configuration options available to a user of a PCS 60, PCS 50, PCS 580, PCS 570/560, 410/400 or PCS 100.

- The Restricted option disables the Users ability to change settings such as Forwarding, Call Waiting, Do Not Disturb etc. On a PCS 60 and PCS 50 the Speed Dials and Ring Tunes tabs are the only tabs that are displayed within the User Configuration/Preferences form. On the PCS 60 the Quick Settings pane is not operational. On a PCS 410/400 the Speed Dials, Ring Tunes and Calibration tabs are the only tabs that are displayed within the User Configuration form. On the PCS 580 only the Group Membership and Speed Dial screens are available when selecting Settings. On the PCS 570/560 only the Logout, Favourites, Groups and the Phone Information options are available within the Settings screen. On the PCS 100 the Forwarding and DND options are removed. This option will also disable manual call recording.
- The Locked option will disable the use of the User Configuration/Preferences form completely on the PCS 60 and PCS 50, and removes the Forwarding and DND options on the PCS 100. On the PCS 60 the Quick Settings pane is not operational. On the PCS 580 the Settings icon is disabled. On the PCS 570/560 the Setting options is removed from the Favourite screen. This option will also disable manual call recording.
- The HotelRoom option is for future use. However, if selected this will be the same as selecting Locked.

The User Display Pref field determines the size of font to be displayed on the User's PCS 410/400, if applicable.

The Contact Mode field determines the editing rights to a Contact entry via their PCS. For further information on Contacts please refer to page 141.

The Edit Mode field enables or disables the User's ability to enter their personal numbers or change their Description field via their PCS 570, PCS 560, PCS 410/400, PCS 60 or PCS 50.   Please refer to the Setting the User's Details section from page 86 for further details.

The Ex Directory option, if ticked, will ensure that this User does not appear in the User Directory on a PCS 410, 400, 100 or 50. This is useful when creating a User that will not be dialled by users of the system.

The Disable System LNR tick box determines whether a User's Last Number Redial list is stored on the call server or locally. By default, this data is stored on the call server. If this option is ticked the data is stored locally on the User's phone but will be lost when their phone is rebooted.

Duplicate User Names and Extension Numbers

It is essential that duplicate User Names and Extension Numbers are not used on a system. To ensure that duplicates have not been entered select Utilities and then Sorted Lists. This program will highlight entries where an identical name or extension number exists. It is also useful for quickly viewing which physical analogue port or PCS a User is assigned to and ensuring that the port or PCS has not been assigned to more than one User.
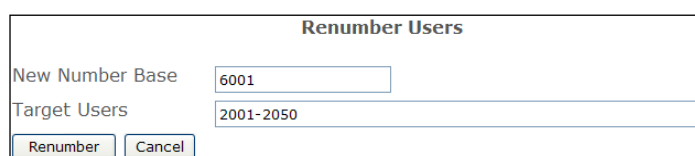
Martin Able	2007 port07,CallServer 00-07-d9-00-30-09	
Marvin Crown	2016 port16,CallServer 00-07-d9-00-30-09	
Mary Winter	2009 port09,CallServer 00-07-d9-00-30-09	
Personnel [Department]	8003	
Personnel Group [Group]		
Rachel Spencer	2010 port10,CallServer 00-07-d9-00-30-09	
Rachel Spencer	2019 PCS 570,CallServer 00-07-d9-00-30-09	*** duplicate ***
Rebecca Woods	2018 PCS 60 Windows PC,CallServer 00-07-d9-00-30-09	
Reception [Department]	8000	
Reception Group [Group]		
Research [Department]	8007	
Restaurant [Department]	8006	

Where multiple Call Servers are being run on one system the User Status facility will verify to which Call Server Users are registered. To view this facility select Utilities and then User Status.

Changing the extension numbers for multiple Users

When the extension number for more than one User is to be changed the Users (Renumber) feature can be used as follows:

1   In Manager, select Users (Renumber)
2   In the New Number Base field enter the first number of the new range to be used, for example, if you wish to renumber the extensions from 6001 to 6050 enter 6001.
3   In the Target Users field enter the extension numbers to be changed. Enter either a comma between each extension number or use a hyphen to indicate a range of extensions. For example: 2001,2003,2005,2009,2015,2020 etc or 2001-2050
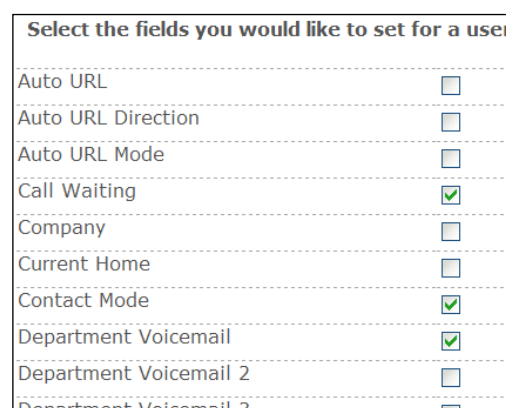
4   Select Renumber
5   The extension numbers for the selected Users will be changed, for example, 2001 will become 6001, and the next extension will become 6002 and so on.
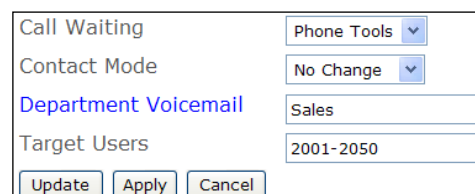
Setting User functionality for multiple Users

You may wish to turn on a particular feature or multiple features for more than one User, rather than turn on these feature in each User's configuration form you can perform this task quickly via the Users (Bulk) feature in Manager.

1   In Manager, select Users (Bulk)
2   Select the feature(s) you wish to configure
3   Select the Continue button at the bottom of the page
4   Select the option required for the relevant field from the list box or tick box provided, or when a field name is displayed in blue the entry for this field can be selected from a list. Click on the field name and select the entry required.

5   In the Target Users field enter the extension numbers for the Users to be changed. Enter either a comma between each extension number or use a hyphen to indicate a range of extensions. For example: 2001,2003,2005,2009,2015,2020 etc or 2001-2050

6   Select Update when ready. The Target Users will be updated with the changes selected.

Music on Hold

For information on assigning music on hold to a specific User please refer to page 179.

Voicemail

For information on voicemail functionality for a User please refer to page 187.

User's Dial Plan

A Dial Plan determines how digits dialled on the User's handset are processed. Therefore a Dial Plan must be assigned to a User to allow external dialling. A Dial Plan can also provide access to PBX functionality via short codes, use of Speed Dials, and be used to allow or bar calls to specific numbers.

1　In Manager select Users
2　Select the User you wish to configure
3　Select the Telephony page
4　Select Dial Plan
5　From the list of Dial Plans displayed select the entry required
6　The selected Dial Plan will be displayed in the Dial Plan field
7　Select Update or Apply when ready.

For further information on working with Dial Plans please refer to page 114.

Displaying a Web Page during a call

A web page can be displayed on a PCS 60, PCS 50 and PCS 410/400 involved in a call to or from a specific User. For example, a member of the Accounts department may require access to accounts records or a member of the Sales team may need access to sales information when dealing with calls to and/or from customers. This web page could be stored on the internal web server (please refer to page 184 for further details), the customer's intranet or accessed externally if an Internet connection is available.

1　In Manager select Users
2　From the User list select the User required.
3　In the Auto URL field enter the path to the web page to be displayed, eg www.splicecom.com/accounts or http://192.168.0.1/sales.php
4　From the Auto URL Mode list box select one of the following:
　　a　NoAction – the web page specified in the Auto URL field can be viewed via the Information icon on a PCS involved in a call to or from this User
　　b　On Ring – the web page specified in the Auto URL field is automatically displayed on a PCS involved in a call to or from this User while the call is ringing
　　c　On Connect – the web page specified in the Auto URL field is automatically displayed on a PCS involved in a call to or from this User once the call is answered
5　From the Auto URL Direction list box select one of the following:
　　a　Bothway – the web page will be displayed on both PCSs involved in an internal call to or from this User. The User's PCS will also display the web page when an external call is either made or received by the User.
　　b　In – the web page will only be displayed on the PCS receiving the internal call either to or from this User. The User's PCS will also display the web page when an external call is received by the User.

c   Out – the web page will only be displayed on the PCS making the internal call either to or from this User. The User's PCS will also display the web page when an external call is made by the User.
6   Select Update or Apply when ready

Displaying a Directory

A list of the Users or Departments on the system or the User's speed dials can be displayed on a PCS 50 or PCS 410/400 when a call is made to and/or received by the User to enable the call to be processed efficiently. (For further information on a User's speed dials please refer to page 102.)
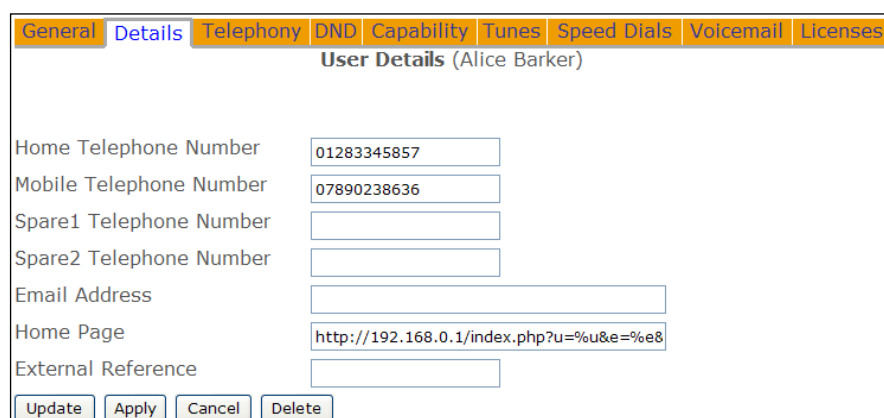
1   In Manager select Users
2   From the User list select the User required.
3   From the Auto URL Mode list box select one of the following:
   a   Favourites – the user's Favourites (Speed Dials) are automatically displayed on a PCS involved in a call to or from this User once the call is answered
   b   Users – the Users Directory is automatically displayed on a PCS involved in a call to or from this User once the call is answered
   c   Departments – the Department Directory is automatically displayed on a PCS involved in a call to or from this User once the call is answered
4   From the AutoURL Direction list box select one of the following:
   a   Bothway – the directory will be displayed on both PCSs involved in an internal call to or from this User. The User's PCS will also display the directory when an external call is either made or received by the User.
   b   Incoming – the directory will only be displayed on the PCS receiving the internal call either to or from this User. The User's PCS will also display the directory when an external call is received by the User.
   c   Outgoing – the directory will only be displayed on the PCS making the internal call either to or from this User. The User's PCS will also display the directory when an external call is made by the User.
5   Select Update or Apply when ready.

Please note: If the User is making a call to or receiving a call for a Department the Auto URL and Auto URL Mode settings for the Department take priority. If the User is making a call to or receiving a call from a Contact the Auto URL and Auto URL Mode settings for the Contact take priority.
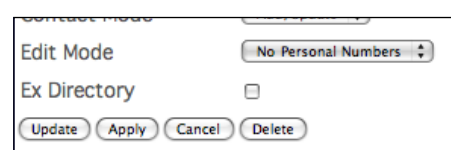
Setting the User's Details

A User's Home and Mobile telephone numbers entered via the Details page, allow other Users to speed dial these numbers from a PCS 580, 570, 560, 410/400, 100, 60 and 50. Spare1 and Spare2 Telephone Number fields can be used to enter two additional numbers if required. (Please note: If the Ex Directory option is ticked this User will not appear in the User Directory available via a PCS 580, 570, 560, 410/400, 100, 60 and 50.)

These numbers will also be matched against incoming CLI and display the User's name in Caller Display. It is therefore very important that extension numbers are **not** entered into these fields. Voicemail can also use these numbers as explained in the Voicemail section from page 187.

Users can also add and edit their own personal numbers via their PCS 580, 570, PCS 560, PCS 410/400, PCS 60 or PCS 50. This facility is controlled by the Edit Mode field under the General Page. By default this is set to All which gives the User full rights to edit his Number fields. However this right can be disabled by selecting No Personal Numbers from the list box.

The No Personal Numbers option will also disable the User's ability to edit his Description field via a PCS 410/400, PCS 60 or PCS 50.

The Home Page field determines which web page will be displayed as default on the User's PCS 410/400, PCS 60 or PCS 50. This could be a page stored on the internal web server (please refer to the Internal Web Server section from page 184 for further details), the customer's intranet or accessed externally if an Internet connection is available.

Using Follow Me

The Follow Me feature will allow a User to take calls, for example, on their mobile or at home, when they are out of the office or at another extension if, for example, the User is spending the day in another part of the building or in another office.

Setting Follow Me

1   In Manager select Users
2   Select the User you wish to configure
3   Select the Telephony page
4   From the Follow Me list box select one of the following options:
    a   None – calls will not be forwarded
    b   Personal – all calls to the User's extension number and DDI will be forwarded.
    c   Dual Personal - all calls to the User's extension number and DDI will be forwarded, and the User's extension will ring. (Calls to the User's Department(s) will ring on the User's extension.)
    d   All - all calls to the User's extension number and DDI will be forwarded plus any calls to a Department to which the User is a member.
    e   Dual All - all calls to the User's extension number and DDI will be forwarded plus any calls to a Department to which the User is a member, and the User's extension will ring.
    f   Personal External - all external calls to the User's extension number and DDI will be forwarded.

g   Dual Personal External - all external calls to the User's extension number and DDI will be forwarded, and the User's extension will ring. (Calls to the User's Department(s) will ring on the User's extension.)

h   All External - all external calls to the User's extension number and DDI will be forwarded plus any calls to a Department to which the User is a member.

i   Dual All External - all external calls to the User's extension number and DDI will be forwarded plus any calls to a Department to which the User is a member, and the User's extension will ring.

j   Personal Internal - all internal calls to the User's extension number and DDI will be forwarded.

k   Dual Personal Internal - all internal calls to the User's extension number and DDI will be forwarded, and the User's extension will ring. (Calls to the User's Department(s) will ring on the User's extension.)

l   All Internal - all internal calls to the User's extension number and DDI will be forwarded plus any calls to a Department to which the User is a member.

m   Dual All Internal - all internal calls to the User's extension number and DDI will be forwarded plus any calls to a Department to which the User is a member, and the User's extension will ring.

5   In the Follow Me To field enter the number to which calls should be routed. This can be an internal extension or an external number eg a mobile number. To forward calls to a Department precede the extension number with a colon eg :8000

6   Select Update or Apply when ready.

User Telephony (Alice Barker)

| Follow Me | Personal |
| Follow Me To | 07890238383 |

Calls to this extension will now follow the User to the number configured and the following will take effect: -

- If the Follow Me To number also has Follow Me set the call will be forwarded on.
- If the Follow Me To number is not answered within the original User's No Answer Time the call will be routed to the original User's voicemail, no further forwarding occurs.
- If the Follow Me To number is not answered within the original User's No Answer Time and the original user has Forward on No Answer set the call will be forwarded to this number.
- If the Follow Me To number is busy the call will be routed to the original User's voicemail, no further forwarding occurs.
- If the Follow Me To number is busy and the original User has Forward on Busy set this will be ignored, the call will be routed to the original User's voicemail, no further forwarding occurs.
- If the Follow Me To number is a Department extension number and a call is not answered and is routed to voicemail this will be the Department's voicemail not the original User's voicemail
- If an external Follow Me To number has a voicemail service, eg a mobile, and this voicemail service answers the call no further forwarding will occur.
- A broken dial tone on the User's handset may indicate that Follow Me has been set.
- Dual Personal and Dual All options are useful for staff that regularly move away from their desks, for example, IT support staff. A call can be answered either at their desk or on, for example, their mobile, whichever is the easiest at the time.

Turn off Follow Me

1   In Manager select Users
2   Select the User you wish to configure

3   Select the Telephony page
4   From the Follow Me list box select None
5   The Follow Me To number can remain for future use
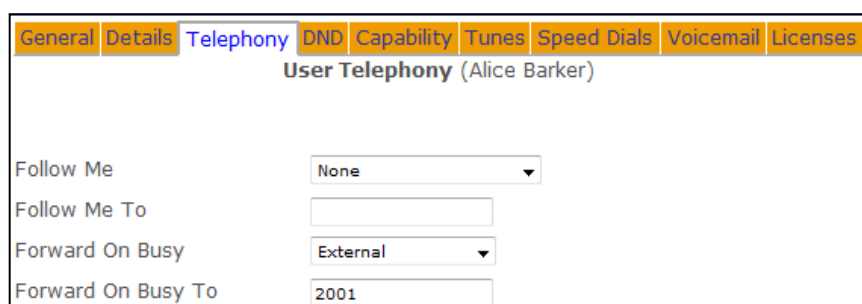6   Select Update or Apply when ready

Please note: The above Follow Me features can be set by the User via their PCS 60, 50, 100, 410/400 or 580/570/560, or via a Dial Plan entry. If a User sets a Follow Me option but does not enter a Follow Me To number the setting will be ignored.

Using Forwarding

The Forwarding feature allows Users to ensure calls are answered when either they are not at their desks or they are busy on another call.

Setting Forward on Busy

1   In Manager select Users
2   Select the User you wish to configure
3   Select the Telephony page
4   From the Forward On Busy list box select one of the following:
    a   Internal&External - all calls to the User's extension number and DDI will be passed to the Forward On Busy To number
    b   Internal –internal calls to the User's extension number and DDI will be passed to the Forward On Busy To number. External calls will be passed to voicemail, if enabled, or will receive the busy tone.
    c   External – external calls to the User's extension number and DDI will be passed to the Forward On Busy To number. Internal calls will be passed to voicemail, if enabled, or will receive the busy tone.
5   In the Forward on Busy To field enter the number to which calls should be routed when this User is busy. This can be an internal extension or an external number eg a mobile number. To forward calls to a Department precede the extension number with a colon eg :8000
6   To activate this feature tick Forward on Busy
7   Select Update or Apply when ready

Calls to this extension will now be forwarded when the User is busy and the following will take effect: -
- If the Forward on Busy number has Follow Me set the call will be forwarded on.
- If the Forward on Busy number is not answered within the original User's No Answer Time the call will be routed to the original User's voicemail, no further forwarding occurs.
- If the Forward on Busy number is busy and Forward on Busy is set for that extension the call will be forwarded on.
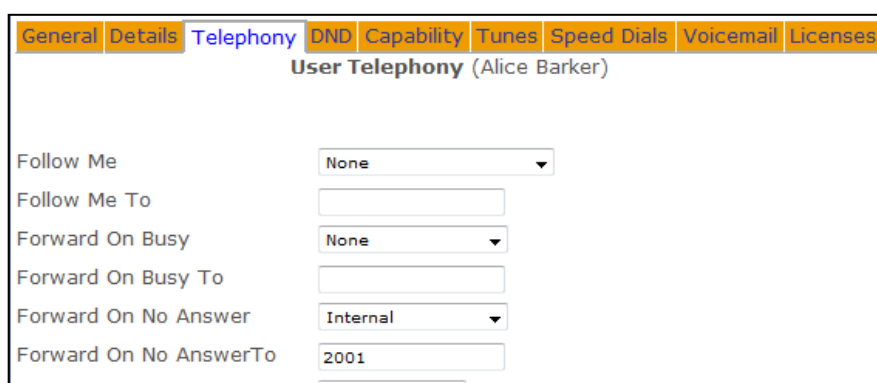
- If the Forward on Busy number is busy and Forward on Busy is not set for that extension the call will be routed to the original User's voicemail.
- If the Forward on Busy number is a Department extension number and is not answered and is routed to voicemail this will be the Department's voicemail not the original User's voicemail
- If an external Forward on Busy number has a voicemail service, eg a mobile, and this voicemail service answers the call no further forwarding will occur.
- A broken dial tone on the User's handset may indicate that Forwarding has been set.

Turn off Forward on Busy

1. In Manager select Users
2. Select the User you wish to configure
3. Select the Telephony page
4. From the Forward on Busy list box select None
5. The Forward on Busy To number can remain for future use
6. Select Update or Apply when ready

Setting Forward on No Answer

1. In Manager select Users
2. Select the User you wish to configure
3. Select the Telephony page
4. From the Forward On No Answer list box select one of the following:
   a. Internal&External - calls to the User's extension number and DDI will be passed to the Forward On No Answer To number if a call is not answered within the User's No Answer Time.
   b. Internal – internal calls to the User's extension number and DDI will be passed to the Forward On No Answer To number if a call is not answered within the User's No Answer Time. External calls will be passed to voicemail, if enabled, if a call is not answered within the User's No Answer Time.
   c. External – external calls to the User's extension number and DDI will be passed to the Forward On No Answer To number if a call is not answered within the User's No Answer Time. Internal calls will be passed to voicemail, if enabled, if a call is not answered within the User's No Answer Time.
5. In the Forward on No Answer To field enter the number to which calls should be routed when this extension is not answered. This can be an internal extension or an external number eg a mobile number. To forward calls to a Department precede the extension number with a colon eg :8000
6. Select Update or Apply when ready.

The amount of time a User's extension will ring before it is considered not answered is determined by the User's No Answer Time field as described below on page 91.

Calls to this extension will now be forwarded after the User's No Answer Time and the following will take effect: -

- If the Forward on No Answer number has Follow Me set the call will be forwarded on.
- If the Forward on No Answer number is also not answered within the original User's No Answer Time the call will be routed to the original User's voicemail, no further forwarding occurs.
- If the Forward on No Answer number is busy and Forward on Busy is set for that extension the call will be forwarded on.
- If the Forward on No Answer number is busy and Forward on Busy is not set for that extension the call will be routed to the original User's voicemail.
- If the Forward On No Answer number is a Department extension number and is not answered and is routed to voicemail this will be the Department's voicemail not the original User's voicemail
- If an external Forward number has a voicemail service, eg a mobile, and this voicemail service answers the call no further forwarding will occur.
- A broken dial tone on the User's handset may indicate that Forwarding has been set.

Turn off Forward on No Answer

1   In Manager select Users
2   Select the User you wish to configure
3   Select the Telephony page
4   From the Forward on No Answer list box select None
5   The Forward on No Answer To number can remain for future use
6   Select Update or Apply when ready

Please note: The above Forwarding features can be set by the User via their PCS 60, 50, 100, 410/400, or via a Dial Plan entry. If a User sets a Forwarding option but does not enter a Forwarding number the setting will be ignored.

Setting the No Answer Time

The No Answer Time is the amount of time the User will be given to answer their extension before the call is routed to voicemail or to their Forward on No Answer number, if configured. It is important to determine the amount of time that will be suitable to the User.   By default, this is set to 20 seconds.

1   In Manager select Users
2   Select the User you wish to configure
3   Select the Telephony page.
4   In the No Answer Time field enter the number of seconds required
5   Select Update or Apply when ready

Using Call Waiting

Call Waiting will allow the User to receive a second call and to toggle between the two calls. When this feature is enabled the User will hear an intermittent beep in the headset when a second call is received.
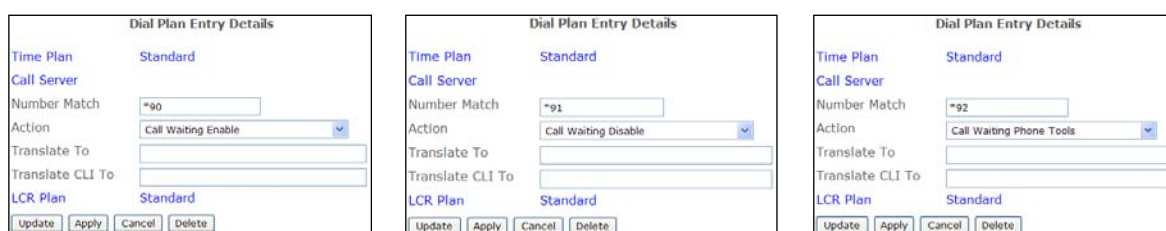
A PCS 60, 50, 410/400 or 580/570/560 will also indicate when a call is waiting and will enable the User to toggle between the two calls.

A User may prefer not to hear the intermittent beep; this can be achieved by selecting the NoBeep option. If a User of a PCS 60 or PCS 50 as a partner to their handset would prefer not to hear the intermittent beep in the headset this can be achieved by selecting the PhoneTools option. The intermittent beep can still be heard via the PCS 60 or PCS 50.

Configuring Call Waiting

1 In Manager select Users
2 Select the User you wish to configure
3 Select the Telephony page
4 From the Call Waiting list box select one of the following options:
    a Disable (default) – the Call Waiting feature is not set for this User.
    b Enable – the User can receive a second call and toggle between the two calls. The User will be informed of a second call by an intermittent beep in the headset and via their PCS 60, 50, 410/400 or 580/570/560.
    c Phone Tools – if the User is running a PCS 60 or PCS 50 as a partner to their handset this option will enable Call Waiting and the intermittent beep will not be heard in the headset but can be heard via the PCS 60 or PCS 50. If the User is not using a PCS 60 or PCS 50 as a partner this option is the same as setting the Enable option.
    d NoBeep – enables Call Waiting but disables the intermittent beep.
5 Select Update or Apply when ready

Users can also set this feature via their PCS 60, 50 or 410/400 (please refer to the relevant User manual for further details) or via a Dial Plan entry similar to the following examples:

Dial Plan Entry Details	Dial Plan Entry Details	Dial Plan Entry Details
Time Plan Standard	Time Plan Standard	Time Plan Standard
Call Server	Call Server	Call Server
Number Match *90	Number Match *91	Number Match *92
Action Call Waiting Enable	Action Call Waiting Disable	Action Call Waiting Phone Tools
Translate To	Translate To	Translate To
Translate CLI To	Translate CLI To	Translate CLI To
LCR Plan Standard	LCR Plan Standard	LCR Plan Standard
Update Apply Cancel Delete	Update Apply Cancel Delete	Update Apply Cancel Delete

The Translate To fields are empty as the Dial Plan entry will only enable or disable Call Waiting for the User on whose handset this short code is dialled. On dialling these short codes a recorded message will be played confirming that Call Waiting has been enabled or disabled.

For further information on working with Dial Plans please refer to page 114.

Call Waiting to an analogue handset can be handled as follows:

1 First call is in progress and the User is informed of a call waiting by the intermittent beep
2 Use the Hold facility to place the first call on hold
3 The second call will be automatically presented
4 Use the Hold facility to end the second call and return to the first caller.

If the User replaces the handset instead of following Step 4 the two calls will be connected together.

(Please refer to the relevant User Manual for details on handling a call waiting via a PCS 60, 50, 100, 410/400 or 580/570/560.)

Call Barring

Users can be barred from making calls to specific numbers or for all external calls.

To bar calls to all external numbers:

1   In Manager select Users
2   Select the User you wish to configure
3   Select the Telephony page
4   Tick Outgoing Call Bar
5   Select Update or Apply when ready

Please note: When this feature is set the User cannot make a call to any external number including emergency services.   Therefore it is important that their Dial Plan contains an entry using the Dial Emergency action to allow the User to dial emergency services. Please refer to the Working with Dial Plans section from page 114 for further details.

To bar calls to specific numbers please refer to the Working with Dial Plans section from page 114 for further details.
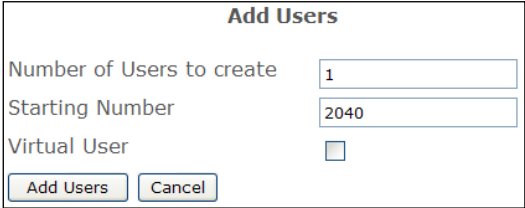
Using Hot Desking

Hot Desking allows Users who may not require a permanent physical extension the ability to use all the facilities provided by the PBX, eg voicemail, forwarding etc, and the ability to log on to a physical phone when in the office, for example. An IPUser licence is required to create a new User on the **max**imiser (please refer to the Licensing section on page 35 for further information).

Hot Desking Users are given a unique extension number and a Login Code to enable them to log on to a physical phone, which could be an analogue handset or a PCS, as and when required. A Login Idle Time can be used to specify how long an extension is idle before that User is automatically logged off.

Creating a Hot Desking User

1   In Manager select Users
2   Select the Add button
3   In the Number of Users to Create field enter *1*
4   In the Starting Number field enter the extension number you wish to give the User, eg 2040.
5   Untick the Virtual User field.
6   Select Add Users
7   You will be returned to the Users List and the new User will be displayed at the top and will be called, eg Extn2040.
8   Select the User and this User can now be configured as required, for example, you may wish to change the User's name and enter a description to further identify this User if required.
9   Select Apply to save any changes made.

User General (James Barrow)

Name	James Barrow
Description	New Sales
Telephone Number	2040
Initial Phone	

10   Select the Telephony page

11   In the Login Access Code field enter a number (max 8 digits) that will be the User's passcode.

12   If required, in the Login Idle Time field, enter the number of seconds this User will be idle before being automatic logged off. (By default this is set to 0 meaning that the User will not be automatically logged off and will need to manually log off.)

Login Access Code	••••
Login Idle Time	0

13   Select Update or Apply when ready

The User's Login Access Code field can contain an asterisk (*). This means any number can be entered during the log in process.

The User will now be able to log on to :-

1   An analogue handset via a Dial Plan entry, similar to the following examples:

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	*2040
Action	Login
Translate To	2040*1234
Translate CLI To	
LCR Plan	Standard

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	##
Action	Logout
Translate To	
Translate CLI To	
LCR Plan	Standard

The Translate to field contains the User's extension number and Login Code.

or

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	*3
Action	Login
Translate To	%r
Translate CLI To	
LCR Plan	Standard

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	*3#
Action	Logout
Translate To	
Translate CLI To	
LCR Plan	Standard

In this example the User must enter
*3 *extension no*login code#*

or

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	*3
Action	Login
Translate To	%r*1#
Translate CLI To	
LCR Plan	Standard

In this example the User's Login Access Code field contains an asterisk (*), therefore the Translate To field must specify the Login Code. The User must enter *3*extension no*.

(For further information on creating Dial Plan entries please refer to the Working with Dial Plans section from page 114.)

2	An analogue handset to which no User is assigned – enter #*extn no*login access code#*, eg #2040*1234#.

3	A PCS 580/560/570, PCS 410/400, PCS 100, PCS 60 IP Phone and PCS 50 IP Phone. (Please refer to the relevant User Manual for details on how to log on to these handsets.)

The handset will return to the assigned User:-
1	When the User logs off via a Dial Plan entry (examples shown above) or via the handset's log out procedure. (Please refer to the relevant User Manual for details on how to log out of the relevant handset.)
2	After the Login Idle Period, or
3	On a reboot of the system

Using the PCS 60 or PCS 50 as a partner when hot desking

Where multiple phones are to be used for hot desking, the PCS 60/50 can be configured to partner which ever handset the hot desking User logs on to. This is useful, for example, for members of staff who, when they come into the office, bring in their laptops and want their PCS 60 or PCS 50 to partner the handset that is available to them.
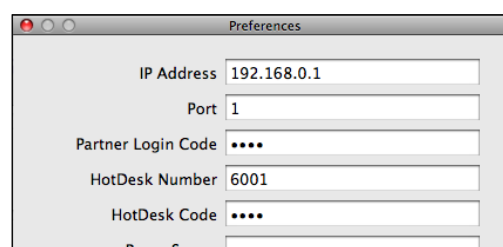
Please note:
• this facility is only available when logging on to a PCS 580/570/560, 100, 410/400 or analogue handset.
• each handset to be used for this facility must be given the same Partner Login Code.

1	For each handset that will be used as a hot desking phone enter a Partner Login Code as follows:
	a	In Manager select Phones
	b	Select the relevant module
	c	Select the port/phone you wish to configure
	d	In the Partner Login Code field enter a 1-8 digit access code. This code must be the same on each phone.

	(For further information on configuring a phone on the system please refer to Configuring an Analogue Extension Port on page 77 and Configuring an IP Phone on page 79.)

2	Open the PCS 60/50 application on the User's PC
3	From the File menu (Windows) or PCS60/50 menu (Mac) select Preferences

4   In the Port field enter any text, eg 1
5   In the Partner Login Code field enter the Partner Login Code configured on each handset as above
6   In the Hotdesk Number field enter the User's extension number
7   In the Hotdesk Code field enter the User's Login Access Code
8   Select OK
9   The PCS 60/50 application will now partner which ever handset the User is logged on to.
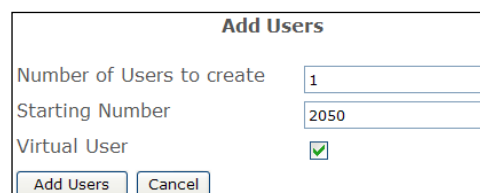
Creating a Virtual User

A Virtual User is a User created on the system to facilitate PBX functionality only. This User does not require a physical extension and will not be logging on to a physical extension. For example, this facility can be used in a school where teachers will be checking and collecting their voicemail from a single phone in the classroom. (For further information on voicemail please refer to page 187.)

An IPVirtualUser licence is required to use this facility. (For further information please refer to the Installing Licences section from page 35.)

1   In Manager select Users
2   Select the Add button
3   In the Number of Users to Create field enter *1*
4   In the Starting Number field enter the extension number you wish to give the User, eg 2050.
5   Ensure that the Virtual User field is ticked.
6   Select Add Users
7   You will be returned to the Users List and the new User will be called, eg Extn2050.
8   Select the User and this User can now be configured as required.
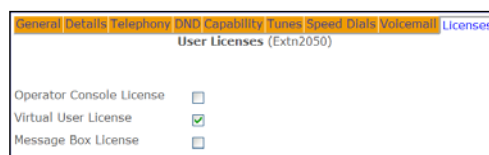
Note that under the Licences page the Virtual User Licence field is ticked

Using Do Not Disturb
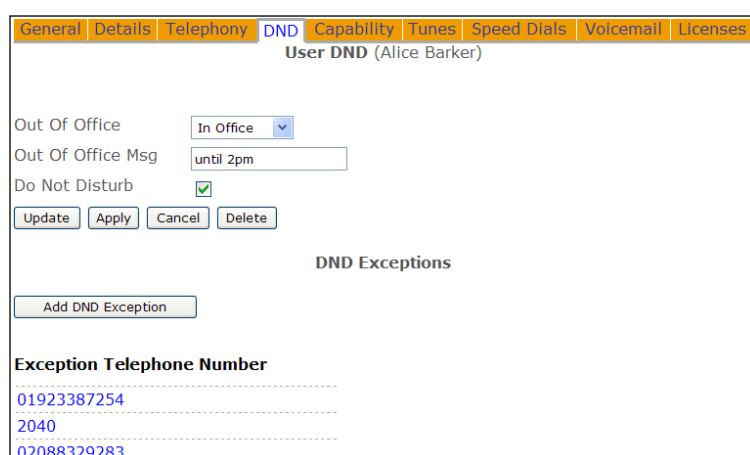
The Do Not Disturb function allows Users to set their phones as permanently busy and a caller will either be given the busy tone, be transferred to voicemail (if enabled) or be transferred to the Forward on Busy number (if set). However, if a User wishes to set Do Not Disturb and also receive a call from a specific number, eg an important call from a customer, this number can be entered as an Exception Telephone Number.

Turn on Do Not Disturb

1   In Manager select Users
2   Select the User you wish to configure
3   Select the DND page
4   Select Do Not Disturb
5   Select Update or Apply when ready

Setting a Do Not Disturb Exception Number

1. In Manager select Users
2. Select the User required
3. Select the DND page
4. Select Add DND Exception
5. In the Telephone Number field enter the internal or external number required
6. Select Update when ready
7. The number entered will be displayed under the Exception Telephone Number list. Click on this number if an amendment is required.
8. Repeat steps 4 and 8 to add all numbers required
9. Select Update or Apply when ready

When a call is received from a DND Exception Number the call will be presented to the extension as normal

The Exception Telephone Number field has the ability to left hand partial match numbers, using ?'s, which means, for example that a DDI range (01923287) or area code (01923) could be matched and the call presented to the extension even though Do Not Disturb is set.

A broken dial tone on the User's handset may indicate that Do Not Disturb has been set.

Users can also set this feature via their PCS 580/570/560, 410/400, 100, 60, 50 or via a Dial Plan entry.

Using an Out of Office Message

Users may wish to inform internal callers when they are out of the office. The message selected will be displayed on the caller's PCS 50, 60, 100, 410/400, 560/570/580. Users can create their Out of Office Message, eg On Holiday until Tuesday, 6th Feb, via their PCS 50, 60 or 410/400; alternatively an Out of Office Message can be configured directly within the User's configuration form.

1. In Manager select Users
2. From the Users list select the User required
3. Select the DND page
4. From the Out of Office list box select the mode required, eg On Holiday.

5   In the Out of Office Message field enter the text to appear with the above selection, eg until Tuesday, 6th Feb.

6   Select Update or Apply when ready
7   The message will appear on a caller's PCS 50, 60, 100, 410/400, 560/570/580 when a call is made to this User's extension (please refer to relevant manual for further details.)
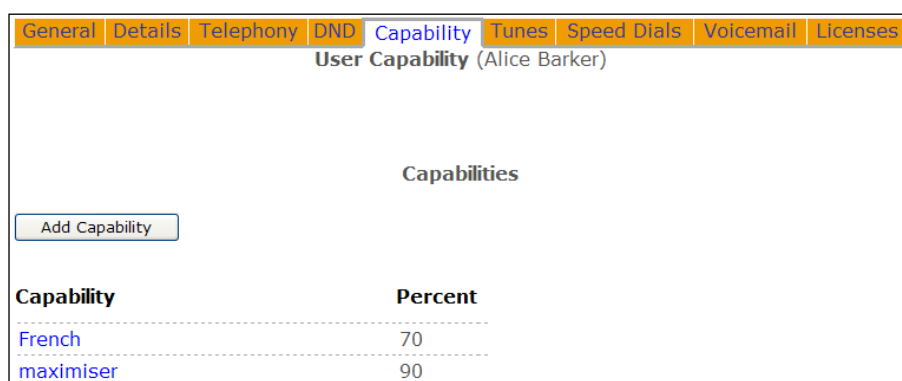
When an Out of Office message has been selected and a caller goes to voicemail the relevant voicemail greeting will be played, please refer to page 199 for further details.

To cancel an Out of Office message select "In Office" from the Out of Office list box. The text in the Out of Office Message field can remain to be used next time this feature is applied.

Setting up Skills Based Routing

A list of skills and the strength of that skill, for example, knowledge of a product for a Help Desk or the ability to speak a foreign language, can be assigned to a User.

1   In Manager select Users
2   Select the User required
3   Select the Capability page
4   Select Add Capability
5   In the Capability field enter the skill, eg French
6   In the Capability Percent field enter the strength of the skill, eg 75
7   Select Update when ready
8   Steps 4 to 7 can be repeated if further capabilities are to be added
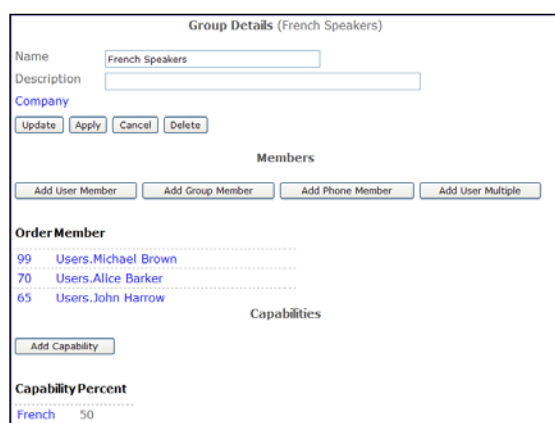
A Group of Users could now be created whereby, for example, all Users must have at least 50% French to answer calls from French customers.

1   In Manager select Groups

2   Click on the Add button
3   In the Name field enter the name required using alpha-numeric characters only, eg French Speakers, and a description if required.
4   Select Apply when ready
5   Select Add Capability
6   Within the Capability field enter the skill required eg French (this must match exactly the name entered in the User's Capability field)
7   Within the Capability Percent field enter the percentage level of skill required eg 50
8   Select Update when ready
9   The Group will be automatically populated with the Users meeting this skill level. Their order number will be the number entered in their Capability Percent field.
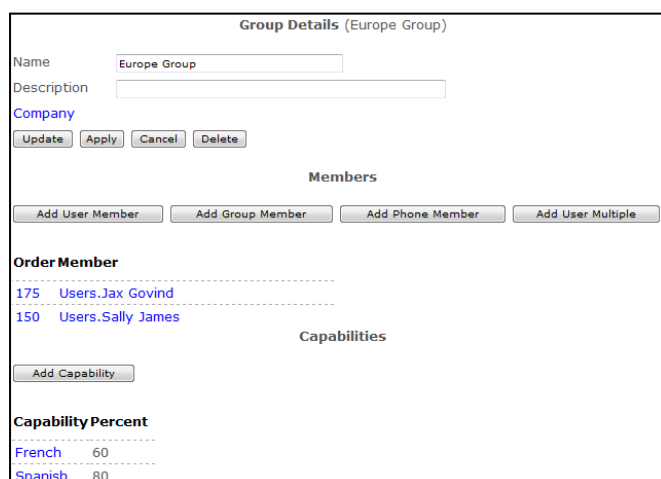
This Group can be assigned to a Department and calls from French customers could be routed to this Department, for example. The order the calls are presented to the members of this Group is determined by the Order field where the higher the number the higher the priority.

If the number of Users who speak French changes the Group or Department does not need to be reconfigured as the Group will automatically look for any Users with a capability configured, for example, at least 50% in French.

If more than one capability is added to a Group, the Group will be populated by Users that meet both criteria, therefore, for example, if a Group requires staff to have 50% fluency in French and 20% fluency in Spanish and you have 10 staff with at least 50% French of which only 5 have at least 20% Spanish then the Group will only have 5 members. Each User's Order number will be the sum of the numbers entered in their relevant Capability percentage fields.
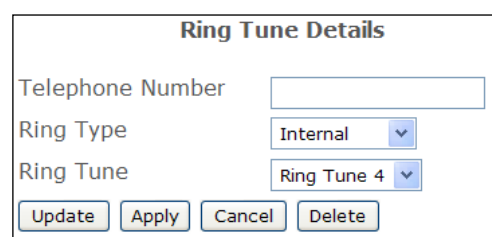
For further information on working with Groups please refer to page 122 and for further information on Departments please refer to page 128.

Changing a User's Ring Tunes

By default, a User will be able to recognise an incoming external call from an incoming internal call by the ring tone. However these default ring tones can be changed for users of a PCS 60 (IP Phone), PCS 50, 100 and 410/400 if required. This can be done by the User via their PCS 60, 50 or 410/400, or alternatively this feature can be configured in Manager as follows:

Change the Ring Tune for an Internal or External Call

1   In Manager select Users
2   Select the User required
3   Select the Tunes page
4   Select Add Ring Tune
5   Delete the entry within the Telephone Number field
6   From the Ring Type list box select either:
    a   Internal – all calls received from an internal extension
    b   External – all calls received from an external number
7   From the Ring Tune list box select the Ring Tune required.
8   Select Update or Apply when ready

Where a User is using a PCS 60 or 50 the Ring Tunes that can be selected via Manager correspond to the WAV files stored on their PC as per this table.

Ring Tunes	Default PCS 60/50 WAV files
Default Internal ring tune	Ring Tune 0.wav
Default External ring tune	Ring Tune 1.wav
Default System ring tune	Ring Tune 2.wav
Ring Tune 3	Ring Tune 3.wav
Ring Tune 4	Ring Tune 4.wav
Ring Tune 5	Ring Tune 5.wav
Ring Tune 6	Ring Tune 6.wav
Ring Tune 7	
Ring Tune 9	

The PCS 60 WAV files are stored by default in C:\Program Files\Splicecom\PCS 60\res. The PCS 50 WAV files are stored by default in C:\Program Files\PCS50\res. All the files are formatted as CCITT A-Law 8.000kHz 8bit mono.

The User's PC must have a sound card in order to use the Ring Tunes facility and the selected ring tune will be player over their PC speakers. Where the PCS 60/50 is configured to partner an analogue handset the analogue phone will still use one of its pre-set ringing cadences.
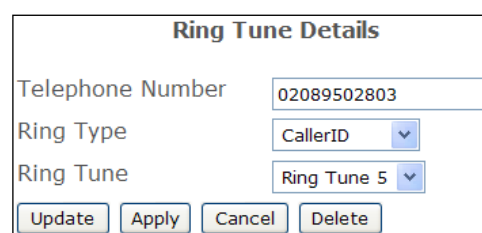
If no ringing is required when using a PCS 50 as an IP phone or as a partner enter a tune number where no corresponding WAV file is stored on the User's PC. If the PCS 50 is partnered to a PCS 100 or 410/400 changing a User's ring tones will change the ringing on the IP handset. Therefore, delete the WAV files stored in C:\Program Files\PCS50\res and the PCS 50 will not ring.

For users of a PCS 570/560 please refer to the "Changing the Logo pic on a PCS 5xx" FAQ on the SpliceCom Forums for instructions on how to set up ring tunes for these phones.

Specify a Ring Tune for calls from a specific number

Calls from a specific external number or internal extension number can be identified by specifying a ring tune for these calls as follows:

1   In Manager select Users
2   Select the User required
3   Select the Tunes page
4   Select Add Ring Tune
5   In the Telephone Number field enter the incoming number to be matched. This can be an internal extension or external number.
6   From the Ring Type list box select CallerID
7   From the Ring Tune list box select the ring tune required.
8   Select Update or Apply when ready
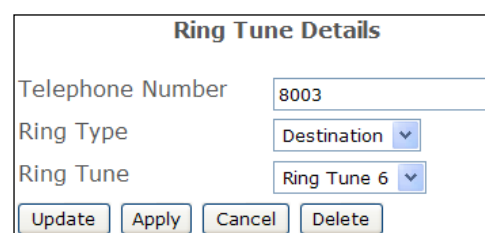
Specify a Ring Tune for personal or Department calls

A User may wish to differentiate between calls to their extension number and DDI number from calls to a Department of which they are a member by using a different ring tune as follows:

1   In Manager select Users
2   Select the User required
3   Select the Tunes page
4   Select Add Ring Tune
5   In the Telephone Number field enter the number to be called. If the User wishes to recognise calls to his extension/DDI enter the User's extension number. If the User wishes to recognise calls to a Department of which he is a member enter the Department's extension number
6   From the Ring Type list box select Destination
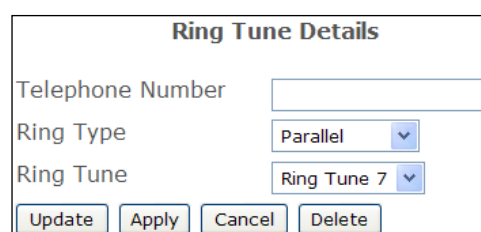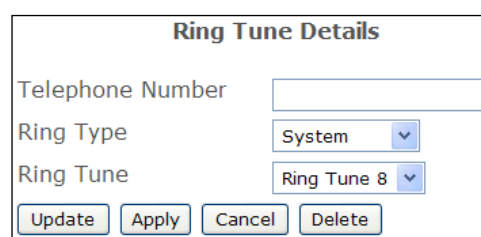7   From the Ring Tune list box select the ring tune required.

8   Select Update or Apply when ready

This entry will take precedence over any other Ring Tune entry.

Specify a Ring Tune when Forward/Follow Me Both Personal is set

When the Follow Me Dual Personal or Dual All feature is set the User's extension will ring as well as the Follow Me To number. (Please refer to page 87 for details on this feature.) The User may wish to have a different ring tune played on their extension to remind them when this feature is set.

1   In Manager select Users
2   Select the User required
3   Select the Tunes page
4   Select Add Ring Tune
5   Delete the entry in the Telephone Number field
6   From the Ring Type list box select Parallel
7   From the Ring Tune list box select the ring tune required
8   Select Update or Apply when ready

Specify a Ring Tune for System calls

System calls are generated by the PBX, for example voicemail call-back. The default ring tune used can be changed as follows:

1   In Manager select Users
2   Select the User required
3   Select the Tunes page
4   Select Add Ring Tune
5   Delete the entry in the Telephone Number field
6   From the Ring Type list box select System
7   From the Ring Tune list box select the ring tune required
8   Select Update or Apply when ready

Working with User Speed Dials

Each User can have a list of regularly used telephone numbers (internal or external) for speed dialling. These can be dialled via a short code or for PCS 580/570/560/410/400/100/60/50 Users via an icon/button displayed in their Favourites list, and for these users a Speed Dial set up for an internal extension will also act as a BLF. A Speed Dial can also be used to pick up a call ringing on that extension.   For information on all these features please refer to the relevant PCS User Manual.

Users of a PCS 60/50, PCS 560/570/580 or PCS 410/400 can create their own Speed Dials; alternatively they can be configured via Manager as follows:

1   In Manager select Users
2   Select the User required
3   Select the Speed Dials page
4   Select Add Speed Dial

5   In the Telephone Number field enter the number (internal or external) to be dialled by the PBX, eg 01923287700

6   In the Short Code field enter the number to be dialled by the User, eg 123. (This entry is not required for a user of PCS 580/570/560/410/400/100/60/50 who wishes to dial via the Speed Dial icon/button only.)

7   In the Description field enter a description for the Speed Dial, eg Head Office.

8   Select Update when ready

9   Repeat steps 4 to 8 to enter further Speed Dials

Speed Dial Details

Telephone Number	01923287700
Short Code	
Description	Head Office

Update   Apply   Cancel   Delete

Please note: User Short Codes take priority over Dial Plan entries.

Disable Call Pick Up

Speed Dials created for internal extensions can also be used to pick up a call ringing on that extension, if however a User on that extension, such as the MD of the company, does not want his/her calls picked up the hash (#) can added to the beginning of the Telephone Number field.

Speed Dial Details

Telephone Number	#2043
Short Code	
Description	Jax

Update   Apply   Cancel   Delete

Please note that users of a PCS 580/570/560 will not see this # in their Settings.

Creating a Speed Dial for multiple Users

When the same Speed Dial is required to be entered for multiple Users this can be done using the Users (Speed Dials) feature as follows:

1   In Manager, select Users (Speed Dial)

2   In the Telephone Number field enter the number (internal or external) to be dialled via this Speed Dial, eg 01313375032

3   In the Short Code field enter the number to dialled by the User, if required, eg 123

4   In the Description field enter text that will identify the Speed Dial, eg Edinburgh office.

5   In the Target Users field enter the extension numbers of the Users to receive this Speed Dial. Enter either a comma between each extension number or use a hyphen to indicate a range of extensions. For example: 2001,2003,2005,2009,2015,2020 etc or 2001-2050

6   Select the Add button
7   The Speed Dial will be created in each of the Target Users' configuration forms within the Speed
      Dial page.

If a particular Speed Dial is to be removed from a group of Users follow the above steps. However at
step 6 select the Remove button instead.

Favourite Groups

Where Users have multiple Speed Dials displayed in their Favourites they may wish to organise these into
groups to make them easier to find. For example, a User may wish to organise her speed dials by project
or department such as Road Show, Directors, Accounts, or separate personal speed dials from business
speed dials, and so on.

This feature can be configured for users of a PCS 580, PCS 570, PCS 560 and PCS 60.

Creating a Speed Dial Group

1   In Manager select Users
2   Select the User required
3   Select the Speed Dials page
4   Select Add Speed Dial
5   In the Short Code field enter GH:*group number*, eg GH:1
6   In the Description field enter a name for the group, eg Exhibition
7   Select Update when ready

Assign a Speed Dial to a Favourite Group

1   Create a new Speed Dial as described above or open an existing Speed Dial
2   In the Short Code field enter G:*group number*, eg G:1
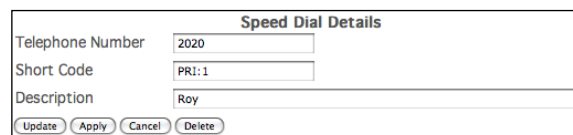3   Select Update when ready.

The Favourite Group [Exhibition] can be selected from the relevant phone's Favourites screen and the Speed Dials assigned to this Group will then be displayed. (Please refer to the relevant User manual for further details.)
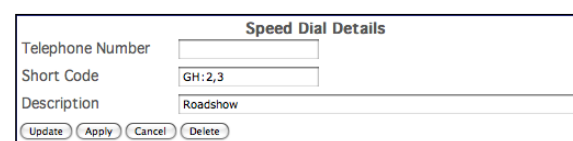
Assigning a display order number to a Speed Dial and Favourite Group

All Speed Dials and Favourite Groups will be displayed in alphabetical order. If however the User wishes to display the entries in a different sequence a display order number can be assigned.
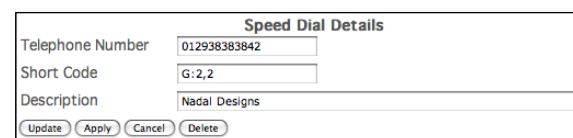
To give a Speed Dial a priority in where it is displayed in the Favourites list enter PRI:*order number*, eg PRI:1, in the Short Code field.

Speed Dial Details	
Telephone Number	2020
Short Code	PRI:1
Description	Roy

Update Apply Cancel Delete

To determine where in the list a Favourite Group will be displayed add a comma and then the order number required, eg GH:2,3.

Speed Dial Details	
Telephone Number	
Short Code	GH:2,3
Description	Roadshow

Update Apply Cancel Delete

To determine where in the list a Speed Dial assigned to a Favourite Group will be displayed add a comma and then the order number required, eg G:2,3.

Speed Dial Details	
Telephone Number	012938383842
Short Code	G:2,2
Description	Nadal Designs

Update Apply Cancel Delete

Where two or more Speed Dials or Favourite Groups are given the same order number these will be displayed in alphabetical order. A Speed Dial or Favourite Group not given an order number will be displayed under those assigned with an order number in alphabetical order.

| General | Details | Telephony | DND | Capability | Tunes | Speed Dials | Voicemail | Licenses |

User Speed Dials (George Kirkland)

Speed Dials

Add Speed Dial

Telephone Number	Short Code	Description
any	GH:1,3	Exhibition
2005	G:1	April
012938383842	G:2,2	Nadal Designs
02082938831	G:1,1	Marwell Exhibitions
2005	G:2,1	Tom
any	GH:2,3	Roadshow
2008	PRI:2	Mark
2020	PRI:1	Roy

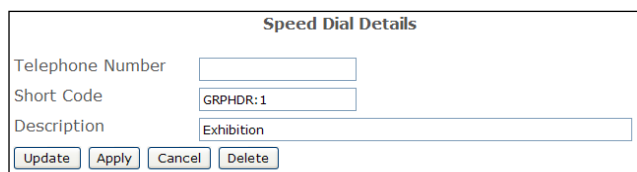This diagram displays an example configuration for a User of a PCS 580, 570 or 560.

Favourite Groups for PCS 100

Where Users have multiple Speed Dials displayed in their Favourites they may wish to organise these into groups to make them easier to find. For example, a User may wish to organise her speed dials by project or department such as Road Show, Directors, Accounts, or separate personal speed dials from business speed dials, and so on.

Creating a Speed Dial Group

1   In Manager select Users

2   Select the User required
3   Select the Speed Dials page
4   Select Add Speed Dial
5   In the Short Code field enter GRPHDR:*group number*, eg GRPHDR:1
6   In the Description field enter a name for the group, eg Exhibition
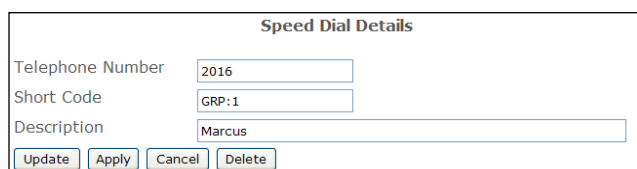7   Select Update when ready

Speed Dial Details

Telephone Number	
Short Code	GRPHDR:1
Description	Exhibition

[Update] [Apply] [Cancel] [Delete]

Assign a Speed Dial to a Favourite Group

1   Create a new Speed Dial as described above or open an existing Speed Dial
2   In the Short Code field enter GRP:*group number*, eg GRP:1
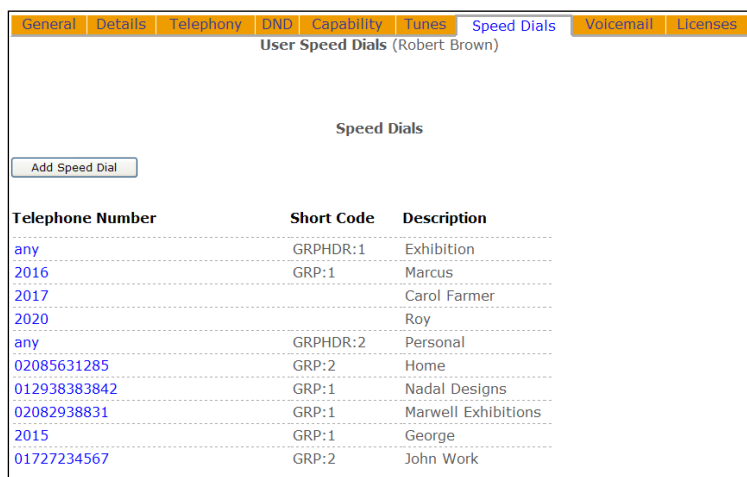3   Select Update when ready.

Speed Dial Details

Telephone Number	2016
Short Code	GRP:1
Description	Marcus

[Update] [Apply] [Cancel] [Delete]

On a PCS 100 the Favourite Group (◉ **Exhibition**) will be displayed in the Favourites screen and can be selected by pressing the Context Sensitive Key beside this entry. The Speed Dials assigned to this Group will then be displayed. (Please refer to the PCS 100 User manual for further details.)

| General | Details | Telephony | DND | Capability | Tunes | Speed Dials | Voicemail | Licenses |

User Speed Dials (Robert Brown)

Speed Dials

[Add Speed Dial]

Telephone Number	Short Code	Description
any	GRPHDR:1	Exhibition
2016	GRP:1	Marcus
2017		Carol Farmer
2020		Roy
any	GRPHDR:2	Personal
02085631285	GRP:2	Home
012938383842	GRP:1	Nadal Designs
02082938831	GRP:1	Marwell Exhibitions
2015	GRP:1	George
01727234567	GRP:2	John Work

This diagram displays an example configuration for a User of a PCS 100.

Using the Park facility

The Park facility on a **max**imiser system allows a call to be put on hold in a system area rather than on the User's extension. This has the benefit of releasing the User's extension to receive further calls. Either the User can pick up the call to continue the conversation or the call can be picked up by another extension on the system. This can be a quick and easy way to transfer calls as per the following examples:

1. A receptionist parks a call on Park 1 and then uses a tannoy system to announce that a call for Mr Bloggs is on Park 1. Mr Bloggs picks up the call wherever he is in the building.
2. A User sits opposite her colleague in the office, she receives a call for her colleague and rather than going through the transfer procedure she can park the call and inform her colleague. Her colleague can then pick up the call on his extension when he is ready.

Calls are parked within a Park Slot and each Park Slot has a number starting from 1. There is no limit to the number of Park Slots that can be used. Therefore calls can be parked in Park Slot 1, 52 or 10012, for example. By default, a User of a PCS 580/570/560, 520, 410/400, 100, 60, 50, or 10 is given access to park slots 1, 2, 3, and 4 via an icon/button (please refer to the relevant User manual for instructions on how to use these park icons/buttons). However, calls can be parked in any Park Slot number and this facility can be used to provide group or private park slots (please see below for further details).

Calls can also be parked and picked up via a Dial Plan entry. An analogue extension without the use of the PCS 60 or PCS 50 can use a Dial Plan entry to utilize this facility. Please see below for further details.

The Park Timeout, in other words the amount of time a call will be parked before it is represented to the User, is set by default to 5 minutes. This time can be changed using the Companies feature as each Company can be configured with its own Park Timeout. Please refer to the Working with Companies section from page 155 for further details.

Each Company uses its own set of Park Slots and therefore a User will not be able to pick up a call parked by a User belonging to different Company. Please refer to the Working with Companies section from page 155 for further details.

Using the Park facility via Dial Plan entries

A call can be parked and picked up via a Dial Plan entry as per the following examples. The call must be put on hold first and then the relevant short code is entered. (Please refer to the Working with Dial Plans section for further details on creating a Dial Plan entry.)
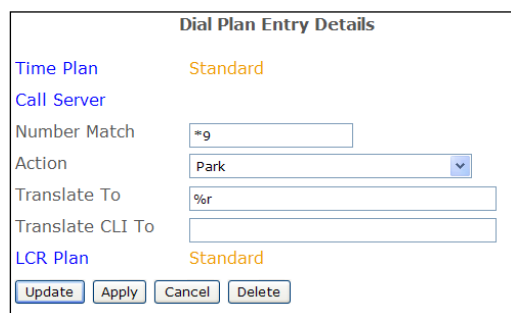
Park a call

The number of the Park Slot in which a call is to be parked is entered in the Translate To field. An entry can be created for each park slot to be used.

The call will be parked in the next available slot starting at 1. The User will be informed via a recorded message which slot the call has been parked in.

Dial Plan Entry Details

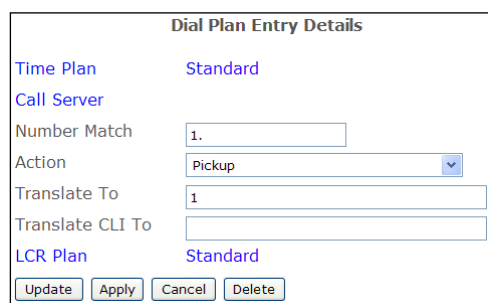|	
Time Plan	Standard
Call Server	
Number Match	*9
Action	Park
Translate To	%r
Translate CLI To	
LCR Plan	Standard

[Update] [Apply] [Cancel] [Delete]

The call will be parked in the slot number entered in the Translate To field. Therefore, in this example, the User would enter *955 to park the call in slot 55.

Pick up a Parked Call

Dial Plan Entry Details
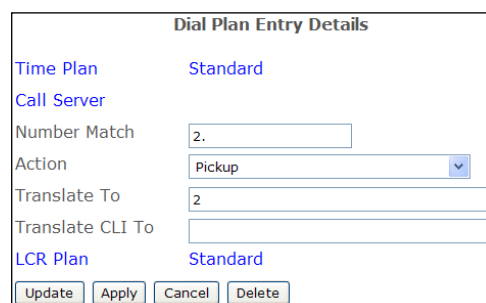
Time Plan	Standard
Call Server	
Number Match	1.
Action	Pickup
Translate To	1
Translate CLI To	
LCR Plan	Standard

[Update] [Apply] [Cancel] [Delete]

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	2.
Action	Pickup
Translate To	2
Translate CLI To	
LCR Plan	Standard
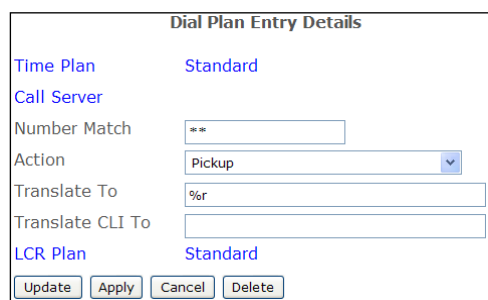
[Update] [Apply] [Cancel] [Delete]

The number of the Park Slot from which a call is to be picked up is entered in the Translate To field. An entry can be created for each park slot to be used.

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	**
Action	Pickup
Translate To	%r
Translate CLI To	
LCR Plan	Standard

[Update] [Apply] [Cancel] [Delete]

The number of the Park Slot from which a call is to be picked up is entered in the Translate To field. Therefore in this example User would enter **55 to pick up a call parked in slot 55.

Changing the Park icons/buttons on a PCS

By default, a User of a PCS 580/570/560, 410/400, 100, 60, or 50 is given access to park slots 1, 2, 3, and 4 via an icon/button (please refer to the relevant User manual for instructions on how to use these park icons/buttons). However a User may wish to have access via one of these icons/buttons to a "private" park slot that is not used by other colleagues. For example, a receptionist may prefer to a park call while she deals with another call without the risk of another user picking up the call; a Sales team may wish to have a "private" park slot for their use only so that they can transfer calls between themselves without the risk of another user picking up the call by mistake. To change the park slots accessed via a User's Park icons/buttons the following can be configured:

1. In Manager select Users
2. Select the User you wish to configure
3. Select the Telephony page
4. In the Park Slot Base field enter the value to be added to the existing park slot numbers, eg 10
5. In the Park Slot Offset field enter the existing park slot number where the re-numbering is to start, eg 2

6   Select Update or Apply when ready.

Dial Plan	Standard
Park Slot Base	10
Park Slot Offset	2
Login Access Code	

In this example the second Park icon/button will now access Park Slot 12, the third accesses Park Slot 13 and the fourth accesses Park Slot 14.

Creating Park icons on a PCS 580, PCS 570, PCS 560 and PCS 60

Rather than changing the park slots that Park icons/buttons 1-4 access as described above, users of a PCS 580, PCS 570, PCS 560 and PCS 60 can be configured with additional Park icons to access any park slots number required. The benefit to this feature is that users can have access to more that the default four Park icons and the Park icons will be displayed with the actual Park Slot number being used, eg Park 99.

This facility is beneficial to a User who wishes to have use of a "private" park slot that is not utilized by other colleagues. Receptionists/operators, for example, who are dealing with a large volume of calls may wish to park a call without the risk of another user picking up the call by mistake. A Sales team, for example, may wish to have a "private" park slot for their use only so that they can transfer calls between themselves without the risk of another user picking up the call.

Users of a PCS 580 or PCS 60 can create these Park icons themselves via Settings; alternatively they can be configured in Manager as follows. For users of a PCS 570/560 these icons can only be created via Manager.

Please note: as a result of creating a new Park icon the icons to Park slots 1-4 will be removed; however they can be re-created as explained below.

1   In Manager select Users
2   Select the User you wish to configure
3   Select the Speed Dials page
4   Select Add Speed Dial
5   In the Telephone Number field enter *slot* following by the slot number required, eg slot8
6   In the Description field enter the text (alpha-numeric characters) you wish to use to identify this Park icon preceded by a colon, eg :Sales Park. If the colon is not entered the Description field is ignored and the icon will be displayed as *Park* followed by the slot number, eg Park 8 (this will follow the same naming convention used on the existing Park icons).

Speed Dial Details

Telephone Number	slot8
Short Code	
Description	Park 8

Update   Apply   Cancel   Delete

or

Speed Dial Details

Telephone Number	slot33
Short Code	
Description	:Sales Park

Update   Apply   Cancel   Delete

7   Select Update or Apply when ready.

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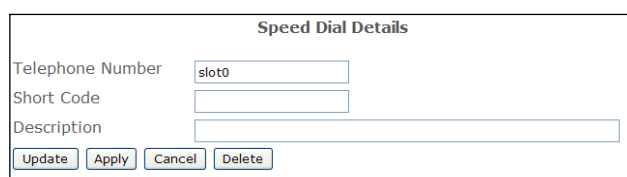8   The new Park icon will be displayed in the User's Favourites screen.

Please note:
- The icons to Park slots 1-4 will be removed; however these icons can be re-created by repeating the above instructions for slot1, slot2, slot3 and slot4.
- If all the Park icons created via a User's Speed Dial page or within Manager are deleted the default Park icons to Park slots 1-4 will return to the User's Favourites screen.

Removing the default Park icons from a PCS 580, PCS 570, PCS 560 and PCS 60

If a User does not wish to use the default Park icons on their PCS 580/570/560/60 and would prefer to use the space for their speed dials these Park icons can be removed. Users of a PCS 580 or PCS 60 can configured this themselves via Settings; alternatively this facility can be configured via Manager as follows. For users of a PCS 570 or 560 this feature can only be configured via Manager.

1   In Manager select Users
2   Select the User you wish to configure
3   Select the Speed Dials page
4   Select Add Speed Dial
5   In the Telephone Number field enter *slot0*.
6   Select Update or Apply when ready.
7   The default Park icons to Park slots 1-4 will be removed from the User's Favourites screen.

	Speed Dial Details	
Telephone Number	slot0	
Short Code		
Description		
Update	Apply	Cancel	Delete

Using Account Codes

Account Codes can be used to identify external calls made for a specific project or client. For example, a code of 1234 can be used to identify an external call made in relation to the new office project being managed by the Office Administrator. This code will be displayed with the Call Logging output, please refer to page 27 for further information.

Firstly, you will need to create the Account Code that the User must dial. This is done via a Contact. For further information on working with Contacts please refer to page 141.

1   In Manager select Contacts
2   Select the Add button
3   In the Name field enter the User's User Name
4   You may wish to tick the Ex Directory field so that this Contact is not displayed within the User Directory on a PCS 580/570/560/410/400/100/60/50.
5   Select Apply when ready
6   Select the Organisation page
7   In the Account (our ref) field enter the code to be dialled by the User, eg 1234
8   Select Update or Apply when ready

A Dial Plan entry must then be created to allow the User to dial the Account Code and make an external call. For further information on working with Dial Plans please refer to page 114.

9   In Manager select Dial Plans
10   Select the Dial Plan being used by the User eg Standard
11   Select Add Dial Plan Entry
12   In the Number Match field enter the digits to be dialled by the User eg *95
13   From the Action list box select Dial Account

14   In the Translate To field enter *%r*
15   Ensure the relevant Time Plan has been selected eg Standard
16   Ensure the entry is pointing to a LCR Plan eg Standard
17   Select Update or Apply when ready

To dial an external number with the Account Code the User must now enter the following on their extension:

Dial Plan Entry Account Code*telephone number
eg *951234*01923289398

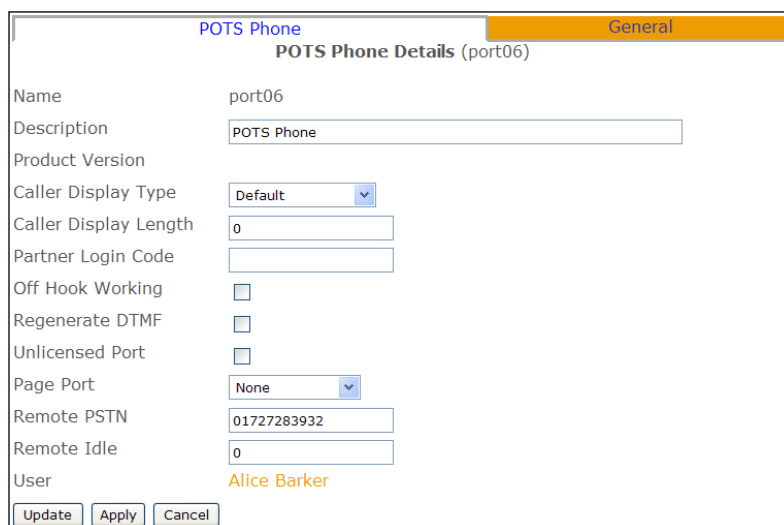Users can be forced to enter an Account Code with every external call as follows:

1   In Manager select Users
2   Select the User required
3   Select the Telephony page
4   Tick the Require Account Coding field
5   Select Update or Apply when ready

Setting up Remote PSTN operation

Remote PSTN operation will allow a User to receive and make calls from, for example, a home telephone or mobile as though they were in the office. This means callers/recipients will be unaware that the User is working from home, the User will not incur any call charges and the User will be able to handle the call as if they were in the office eg transfer calls. The **max**imiser will send CLI information with all calls so that if the User is able to receive CLI at home or on their mobile (this service is arranged through the User's telephone company) the User will be able to differentiate between business and personal calls. No drop in speech quality will occur when using PSTN lines which may happen when using VoIP over the Internet.

This feature is available for a user of a PCS 580/570/560 or an analogue phone port and is configured within the phone's or port's configuration form as follows:

1   In Manager select Users
2   Select the User that requires the Remote PSTN feature
3   Select the entry within the Initial Phone field, eg Phone Module 1.port06
4   You will be taken to the phone or port assigned to this User
5   Within the Remote PSTN field enter number required, for example, the User's home or mobile telephone number.
6   Select Update or Apply when ready (if you select Update you will be returned to the User's configuration form.)

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POTS Phone	General

POTS Phone Details (port06)

Name	port06
Description	POTS Phone
Product Version	
Caller Display Type	Default
Caller Display Length	0
Partner Login Code	
Off Hook Working	☐
Regenerate DTMF	☐
Unlicensed Port	☐
Page Port	None
Remote PSTN	01727283932
Remote Idle	0
User	Alice Barker

Update   Apply   Cancel

The User assigned to this phone or port will now receive calls on the number configured, for example, on their home or mobile telephone, as though they were in the office. Internal extensions can dial the User's extension number and the home or mobile phone will ring.

A mobile phone can transfer a call by pressing the * key three times which provides the dial tone and the destination number can be entered. Once the call has been answered the original call can be transferred by ending the call on the mobile. Please note that the caller will hear the tones when the * is pressed.

Please note:
- The phone or analogue phone connected to this port will now be unusable.
- Ensure that Follow Me has not been configured for this User as two calls will be received at the number configured. If the Follow Me call is answered the User will be unable to use Remote PSTN functionality eg they will be unable to transfer the call.
- The system will make one attempt to route the call to the Remote PSTN number. If the number is busy or not answered the call will be handled in the usual manner eg routed to voicemail if configured.

To enable a User to make calls from, for example, their home telephone number and not incur call charges outgoing calls must be made via a PCS 60/50 set up as a partner to the User's phone or analogue port (please refer to the PCS 60 section from page 58 for further details). The PCS 60/50 must be running on a PC with a VPN connection to the office.

When a call is made via the PCS 60/50 in addition to a call being made to the destination the **max**imiser will also ring the User's Remote PSTN number therefore the office is billed for both parts of the call and the User's office DDI number will be sent with the CLI information rather than the User's home number.

Please note: PCS 60/50 only needs to be running when an outgoing business call is to be made.

When a call is received a typical home telephone (analogue PSTN) will not clear the call when the phone is put down. Therefore an outgoing call made via the PCS 60/50 must be ended via the PCS 60/50 to clear the call from the **max**imiser. In case a User forgets to end a call via the PCS 60/50 the system will automatically end the call after the 60 seconds default idle period. However the PCS 60/50 must be running for this idle period to be activated. The default idle period can be configured as follows:

1   In Manager select Users

2   Select the User that requires the idle period to be changed

3   Select the entry in the Initial Phone field eg Phone Module 1.port06

4   You will be taken to the phone or port assigned to this User

5   Within the Remote Idle field enter the number of seconds required.

6   Select Update or Apply when ready (if you select Update you will be returned to the User's configuration form.)
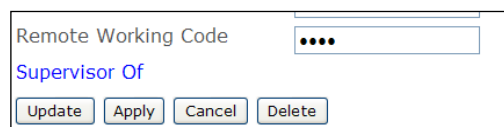
Please note:

- It may be useful to have a home telephone that has loudspeaker facilities so that the user can hear calls that ring during the idle period and when the local line clears.
- A mobile phone is able to clear incoming calls itself.

Remote PSTN operation is not configurable for a user of a PCS 410/400/100. If this feature is required assign the User to an analogue phone port and configure as above. When the User is in the office and wishes to use a PCS 410/400/100 the User should log on to this phone and this will disable the Remote PSTN feature (please refer to Using Hot Desking on page 93 for further details). When the User logs off from the PCS 410/400/100 the User will be returned to the assigned analogue phone port and therefore the Remote PSTN feature will be enabled.

Setting Remote PSTN via a PCS 580, PCS 570, PCS 560 or PCS 60

Users' of a PCS 580, 570, 560 and 60 can activate the Remote PSTN feature via their phones. Firstly the User must be given permission to use this facility as follows. You will need the Partner Login Code used by the User's phone.

1   In Manager select Users

2   Select the User required

3   Select the Telephony tab

4   In the Remote Working Code field enter the Partner Login Code used by this User's phone.

5   Select Update or Apply when ready.

A user of a PCS 60 will now be able to turn Remote PSTN on or off via the Extension Anywhere list box displayed in the Quick Settings bar. (Please refer to PCS 60 user manual for further details.)

Users of a PCS 580, 570 or PCS 560 will be able to turn Remote PSTN on or off via Settings. (Please refer to the relevant user manual for further details.)

If numbers have been entered in the Details page of their User configuration form the User will be given the option to select these numbers from a ready-made list. Alternatively, a new number can be entered.

When the User selects or enters the number required this will be copied into the Remote PSTN field within their phone's or analogue port's configuration form, thereby enabling the Remote PSTN feature.

When the User selects None, the number within their Remote PSTN field is removed, thereby disabling the Remote PSTN feature.

Working with Dial Plans

Dial Plans determine how digits dialled on a User's extension are handled. A Dial Plan will allow you to create speed dials, short codes to system features and set up call barring. A Dial Plan can be assigned to one or many Users. Multiple Dial Plans can be created to allow or disallow certain functionality for specific Users or where Users belong to different Companies (please refer to the Working with Companies section from page 155 for further details).

Creating a Dial Plan

1   In Manager select Dial Plans
2   Click on the Add button
3   Enter a Name, using alpha-numeric characters only, do not use any punctuation and do not begin with a number.
4   Enter a Description that will help you to identify why you have created this Dial Plan, if required
5   In the Dial Delay Time field enter the amount of time (in milliseconds) after digits are entered before the system starts to process the number dialled.
6   In the Dial Delay Count field the number of digits to be entered before the system starts to process the number dialled.
7   Select Update or Apply when ready
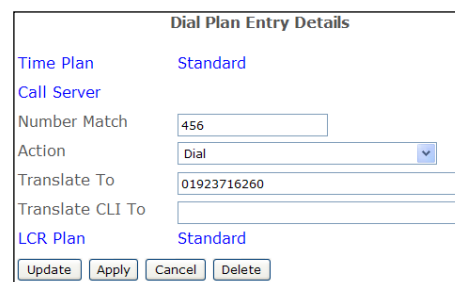
Creating a Dial Plan Entry

1   Open the relevant Dial Plan
2   Click on Add Dial Plan Entry
3   If this entry is to be active only during certain hours click on Time Plan and select the Time Plan required. By default the Standard Time Plan is used. (See the Using a Time Plan section from page 148 for further information).
4   In the Number Match field enter the digits to be dialled on an extension eg 456. The digits dialled are matched from the left and the system will look for closest match. Therefore a Dial Plan entry of 2 and 256 can exist in the same Dial Plan. This field can contain digits 0-9, * and #.

5   From the Action list box select the action to be performed by the system eg Dial. (Please refer to page 294 for a description of each Action.)

6   In the Translate To field enter the digits to be processed by the system eg 01923716260

7   If you wish to specify the outgoing CLI to be sent when this Dial Plan entry is used enter the User or Department's extension number in the Translate CLI To field.
(Please refer to the Force Outgoing CLI section from page 118 for further details.)

8   If you wish to assign this entry to a specific LCR Plan click on LCR Plan and select the plan required. By default the Standard LCR Plan is used. (See page 175 for further information.)

9   Select Update when ready

10   Repeat steps 7 to 14 until all the required short codes, speed dials etc have been entered

Dial Plan Entry Actions

A full list of the Actions available and their descriptions are available within Field Descriptions section from page 294.

Standard Dial Plan

By default, a Dial Plan called Standard is created which, when assigned to a User, allows the User to make external calls (no prefix is required), strips the prefix of 9 from a number dialled for those who are used to dialling 9 for an outside line, allows emergency calls to be made even when outgoing call barring is set and provides the short code of 1571 to allow access to the User's voicemail box. Additional Dial Plan entries can be created within this Dial Plan by following steps above.

A ready-made list of Dial Plan entries is available with the **max**imiser system and can be imported into the configuration. For further details please refer to the Importing and Exporting the Configuration section from page 242.

Dial Plan Commands

The following commands can also be used within a Dial Plan entry:

In the Number Match field –
? (question mark) = wild card entry, to match the number entered
. (full stop) = the digits dialled by the user will match exactly the digits entered before the full stop. This means, for example, that a Dial Plan entry of 1. can be used but still allow Users the ability to dial 100 to BT.

In the Translate To field -
%n = dial all digits entered by the user
%r = dial the number entered by the user minus the digits entered in the Number Match field
! (exclamation mark) = go to voicemail (please refer to page 187 for further details)
-n (where n is a number) = used in conjunction with the Park action, instructs the system to park the call in the next available slot starting at n, eg -1
%e = dial the original extension (mainly used in a LCR Plan entry to allow extension level billing but still withhold CLI, for example, via the Dial Plan.)
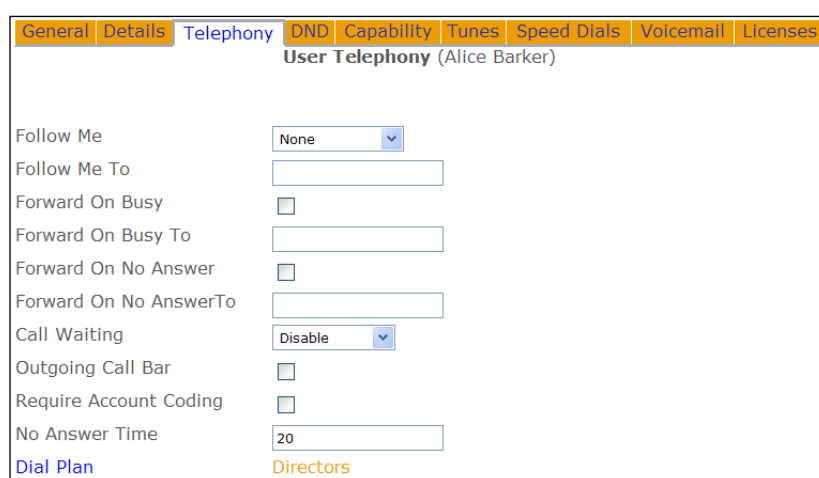
In the Translate CLI To field -

? (question mark) = withhold the number

Assigning a Dial Plan to a User

Dial Plans are assigned to Users and determine how digits dialled on their extension will be handled.

1   In Manager select Users
2   Select the User required
3   Select the Telephony page
4   Select Dial Plan
5   Select the Dial Plan required and this will be displayed in the Dial Plan field
6   Select Update or Apply when ready

A User will require a Dial Plan in order to make outgoing calls therefore if the User's Dial Plan field is blank this field will default to the Standard Dial Plan. It is therefore recommended that default Standard Dial Plan is not deleted.

Using a Common Dial Plan

If different Dial Plans are going to be used by different Users and there are common entries in each dial plan these common entries can be created in one Dial Plan and then associated with another Dial Plan so that both Dial Plans are used. This means that these common entries do not need to be stored in each Dial Plan.

1   Create a Dial Plan and create the entries that will be common to each User.
2   Create another Dial Plan and create the entries that will be specific to the particular User(s)
3   Within this Dial Plan select Common Dial Plan
4   From the Dial Plan list select the Dial Plan containing the common entries
5   Select Update or Apply when ready

Dial Plan Details (Directors)

Name	Directors
Description	
Dial Delay Time	1500
Dial Delay Count	4
Common Dial Plan	Standard

[Update] [Apply] [Cancel] [Delete]

Please note:

- The system will check for a match within the Common Dial Plan first and then check the Dial Plan assigned to the User.
- The Dial Delay Time and Dial Delay Count will be read from the Dial Plan assigned to the User.
- Dial Plans can be cascaded up to 20 levels - regardless of the number of Dial Plans that have been cascaded the system will check the Dial Plan at the bottom first before working to the top, ie the Dial Plan assigned to the User, and the User's Dial Delay Time and Count will affect every entry regardless of which Dial Plan it is in.

Creating a Dial Plan entry to allow External calls

A Dial Plan determines how digits dialled on the User's handset are processed. Therefore a Dial Plan must be assigned to a User to allow external dialling. The following Dial Plan entries give examples on how to allow external dialling:

Dial Plan Entry Details

Time Plan	
Call Server	
Number Match	
Action	Dial
Translate To	%n
Translate CLI To	
LCR Plan	Standard

[Update] [Apply] [Cancel] [Delete]

In this example the Number Match field is left blank so that all numbers dialled, unless matched by an existing Dial Plan entry, will be routed via this Dial Plan entry.

Dial Plan Entry Details

Time Plan	
Call Server	
Number Match	9
Action	Dial
Translate To	%r
Translate CLI To	
LCR Plan	Standard

[Update] [Apply] [Cancel] [Delete]
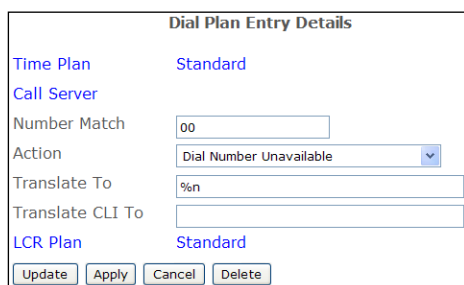
In this example a prefix is required to allow external dialling.

Setting Call Barring

As well as allowing external calls to be made Dial Plan entries can be used to disallow external calls.

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	00
Action	Dial Number Unavailable
Translate To	%n
Translate CLI To	
LCR Plan	Standard

[Update] [Apply] [Cancel] [Delete]
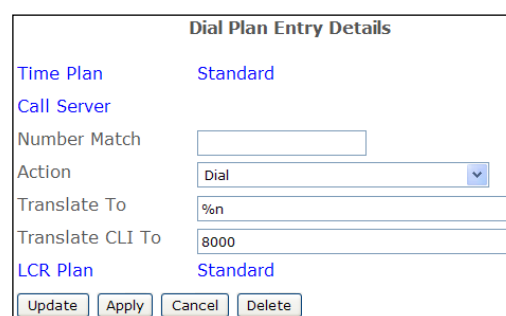
In this example calls to international numbers are barred.

Forcing Outgoing CLI

The Translate CLI To field in a Dial Plan entry can be used to specify the outgoing CLI for a specific call. This is useful if you wish, for example, the main company number to be presented with external calls rather than the User's individual DDI number.

A User or Department's Extension Number should be entered in the Translate CLI To field similar to this example.

This number will be used as the CLI for all internal calls. For external calls the User or Department's DDI number as specified in the relevant DDI Call Plan will be presented as the CLI. (Please refer to Working with DDI Plans section from page 169 for further information.)

Dial Plan Entry Details
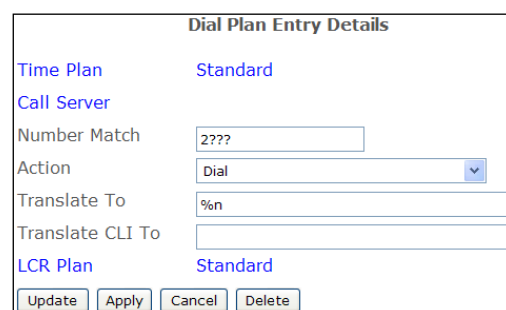
Time Plan	Standard
Call Server	
Number Match	
Action	Dial
Translate To	%n
Translate CLI To	8000
LCR Plan	Standard

[Update] [Apply] [Cancel] [Delete]

If the Users' own extension numbers are to be used as CLI for internal calls this example Dial Plan entry for each extension number range can be created.

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	2???
Action	Dial
Translate To	%n
Translate CLI To	
LCR Plan	Standard

[Update] [Apply] [Cancel] [Delete]

An entry in the Translate CLI To field will take precedence over a User's DDI Call Plan entry (please refer to page 165 for further information) and the Trunk's Presentation Number (please refer to page 159 for further information).

Example Dial Plan Entries

The following diagrams give examples Dial Plan entries that can be created to make use of features available on the system. Further examples are available within the relevant sections of this manual.

A speed dial

An external number, extension number, User Name or Department Name can be entered in the Translate To field.

Override the Outgoing Call Bar option

The external number to be dialled is entered in the Translate To field.

Withhold CLI
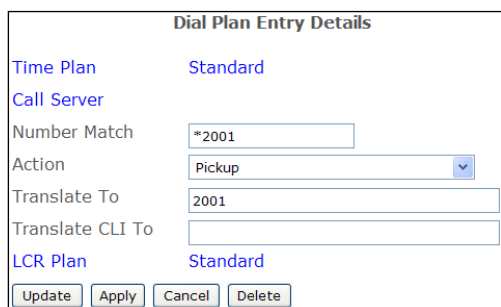
In this example Users can prefix a number with 141 if they wish to withhold their CLI.

Pick up a call to a User or Department
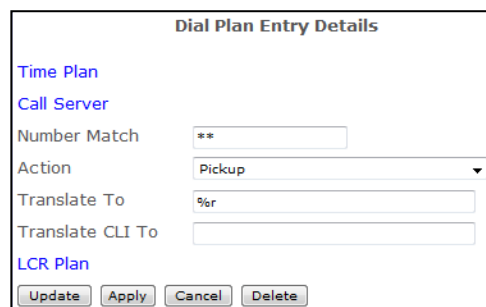
The Extension number, User Name or Department Name required is entered in the Translate To field.
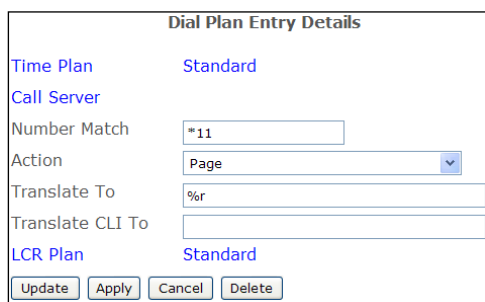
In this example if an extension number is not entered after the asterisks (*) the longest waiting call on the system will be picked up.

Please note: if the Translate To field contains a number of two digits or less this will translate to a Park Slot number.

Page an extension

The extension number of the phone to be paged is entered in the Translate To field. Therefore in this example the User would enter, for example, *112030#.

(For details on how to set up an extension's analogue port to use an external paging source, for example, a tannoy, please refer to page 78.)

Activate an External Relay port
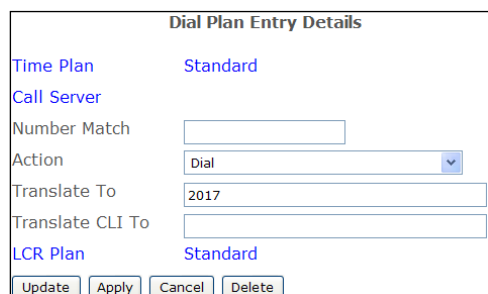
The Translate To field must contain the External Relay port to be activated, either 1 or 2.

Automatically dial an extension when a specific phone is picked up

Dial Plan Entry Details

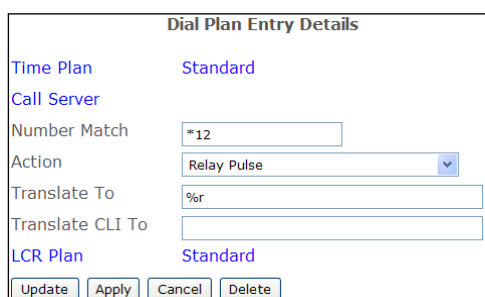|	
Time Plan	Standard
Call Server	
Number Match	
Action	Dial
Translate To	2017
Translate CLI To	
LCR Plan	Standard

Update Apply Cancel Delete

The extension number of the phone to be picked up is entered in the Translate To field.

Create this entry within a new Dial Plan, set the Dial Delay Count and Time to 0 and configure the User assigned to the phone with the new Dial Plan.

Silent call monitoring

Dial Plan Entry Details

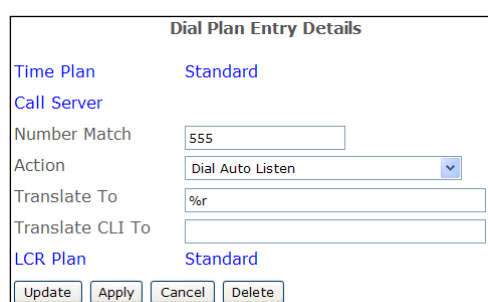|	
Time Plan	Standard
Call Server	
Number Match	555
Action	Dial Auto Listen
Translate To	%r
Translate CLI To	
LCR Plan	Standard

Update Apply Cancel Delete

The extension number to be monitored is entered in the Translate To field. Therefore, in this example the User would dial, for example, 5552042.

Default Dial Plan entries

Files containing ready-made Dial Plan entries are available with the system and can be imported into configuration as described in the Importing and Exporting the Configuration section from page 240. These files also contain the Dial Plan entries required for use with the pre-programmed buttons on the PCS 10 and PCS 520 analogue handsets, and the Dial Plan entries required to match the System Feature Guide displayed on the PCS 5 and PCS 505. Dialplan.txt contains the entries required for the PCS 5, PCS 10, PCS 505 and PCS 520. Dialplan520.txt contains the entries required for the PCS 520. These entries can be created manually if preferred.

Dial Plan entries required for the PCS 520

Number Match	Action	Translate To
*50	Voicemail Retrieve	
*51	Park	-1
*52	Pickup	1
*53	Pickup	2
*54	Pickup	3
*55	Pickup	4
56	Follow Me Set	%r
*57	Follow Me Enable	
*58	Follow Me Disable	
*59	Voicemail Greeting	

Dial Plan entries required for the PCS 10 and PCS 5

Number Match	Action	Translate To
1571	Voicemail Retrieve	
##.	Park	-1
1.	Pickup	1
2.	Pickup	2
3.	Pickup	3
4.	Pickup	4
21	Follow Me Set	%r
*210	Follow Me Disable	
*211	Follow Me Enable	

Dial Plan entries required for the PCS 505

Number Match	Action	Translate To
##.	Park	-1
1.	Pickup	1
2.	Pickup	2
3.	Pickup	3
4.	Pickup	4
*50	Voicemail Retrieve	
56	Follow Me Set	%r
*57	Follow Me Disable	
*58	Follow Me Enable	
*59	VoicemailGreeting	

Working with Groups

A Group can be used to arrange together a particular selection of Users, Phones and/or Groups.   A Group could contain Users with a similar job function and are required to answer specific calls, or contain certain Phones that are to receive calls regardless of who is logged onto the handsets, for example.
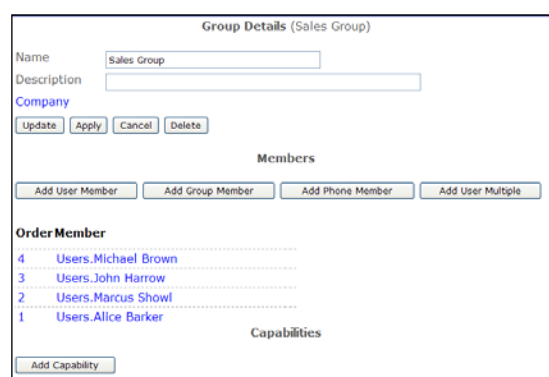
Creating a Group of Users

A Group can be used to link together a set of Users, for example, by department, job function or by skill level, eg the ability to speak a foreign language.

By default, a Group called MainGroup is created containing the first two Users eg Extn2001 and Extn2002.

1   In Manager select Groups
2   A list of the current Groups will be displayed
3   Click on the Add button
4   In the Name field enter the name to be used for the Group. Use alphanumeric characters only, begin with a letter and do not use any punctuation.
5   If required, in the Description field enter text that will help to identify the Group
6   Select Apply
7   Click on Add User Member
8   Click on the Member field

9   From the Select User list select the User you wish to add and this name will appear in the Member list

10 In the Order field enter the number required. The Order number determines the priority of the member within the Group. The higher the number the higher the priority, for example, a User with an Order number of 4 will have a higher priority than a User with an Order number of 1. The member with the higher number will be presented with a call first, then the next highest and so on.

11 Select Update when ready and you will be returned to the Group.

12 Repeat steps 7 to 11 to add further members. Each new member will be given the next Order number. To change a member's Order number enter the required number in the Order number field. The Members will be listed in Order number sequence with the highest at the top.

13 Select Update or Apply when ready.

For information on how a Group is used to present calls please refer to Routing Calls via a Department on page 128.

Change a User member

1   In Manager select Groups
2   A list of the current Groups will be displayed
3   Select the Group required
4   Click on the member to be amended
5   From the Select User list select the User required and you will be returned to the Group
6   Select Update or Apply when ready.

Changing the Order number for a member of a Group

1   In Manager select Groups
2   A list of the current Groups will be displayed
3   Select the Group required
4   Select the Order number for the member to be amended
5   In Member Details form within the Order field enter the number required.
6   Select Update when ready and you will be returned to the Group.
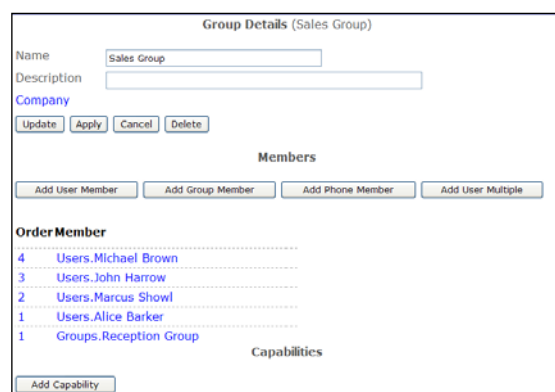7   Select Update or Apply when ready.

Delete a Member from a Group

1   In Manager select Groups
2   A list of the current Groups will be displayed
3   Select the Group required
4   Select the Order number for the member to be deleted
5   In the Member Details form select Delete and you will be returned to the Group
6   Select Update or Apply when ready.

Adding a Group to a Group

Where the Users to be added to a Group are already contained in an existing Group, rather than entered each User in separately the existing Group can be added instead.

1   In Manager select Groups
2   Select the Group required
3   Select Add Group Member
4   Select the Member field
5   From the Select Group list select the Group required.
6   You will be returned to the Member Details form
7   In the Order field enter the number required. The Order number determines the priority of the member within the Group.   The higher the number the higher the priority, for example, a User with an Order number of 4 will have a higher priority than a User with an Order number of 1. The member with the higher number will be presented with a call first, then the next highest and so on.Select Update when ready
8   The Group will appear in the Members list
9   Repeat steps 3 to 9 to add further Groups. Each new member will be given the next Order number. To change a member's Order number enter the required number in the Order number field. The Members will be listed in Order number sequence with the highest at the top.
10   Select Update or Apply when ready.

A Group member can be deleted and amended in the same manner as described for a User member above.

Creating a Group of Phones

A Group can be used to link together a set of Phones. This Group can then be used, for example, to route calls to these handsets or page these handsets or send an Event to the handsets.   The benefit of this facility is that a call/message will be sent regardless of who is logged on to the handsets at the time.

For further information on Events please refer to the Using Events section from page 181.

1   In Manager select Groups
2   A list of the current Groups are displayed
3   Click on the Add button
4   In the Name field enter the name to be used for the Group. Use alphanumeric characters only, begin with a letter and do not use any punctuation.
5   If required, in the Description field enter text that will help to identify the Group
6   Select Apply
7   Select Add Phone Member
8   Select the Member field
9   From the Select Module for Phones list select the Module to which the phones are connected

10   From the Select Phone list select the phone required and this will appear in the Member list.In the Order field enter the number required. The Order number determines the priority of the member within the Group.   The higher the number the higher the priority, for example, a User with an Order number of 4 will have a higher priority than a User with an Order number of 1. The member with the higher number will be presented with a call first, then the next highest and so onSelect Update when ready and you will be returned to the Group.

11   Repeat steps 7 to 12 can add further members. Each new member will be given the next Order number. To change a member's Order number enter the required number in the Order number field. The Members will be listed in Order number sequence with the highest at the top.
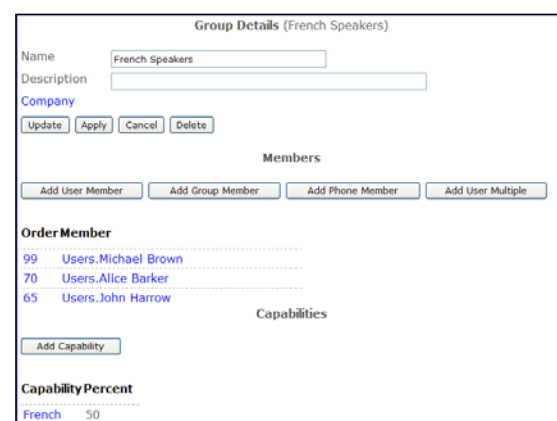
12   Select Update or Apply when ready.

A Phone member can be deleted and amended in the same manner as described for a User member above.

Creating a Group of Users with specific skills

A Group can be set up to be automatically populated with User's who meet a specific skill, for example, those that can speak French, or those that can support the **max**imiser etc. To set up a User's Capability please refer to Setting up Skills Based Routing on page 98.

1   In Manager select Groups

2   A list of the current Groups will be displayed

3   Click on the Add button

4   In the Name field enter the name to be used for the Group. Use alphanumeric characters only, begin with a letter and do not use any punctuation.

5   If required, in the Description field enter text that will help to identify the Group

6   Select Apply

7   Select Add Capability

8   In the Capability field enter the skill required, eg French. This must match exactly the Capability entered in the User's Capability page.

9   Within the Capability % field enter the percentage level of skill required eg 50

10   Select Update and you will be returned to the Group

11   The Group will be automatically populated with the Users meeting this skill level. The Order field will display the User's Capability %. This will also determine the order of the Group where the higher the number the higher the priority.
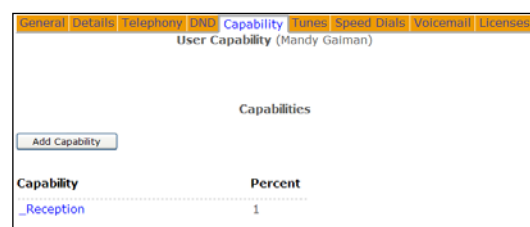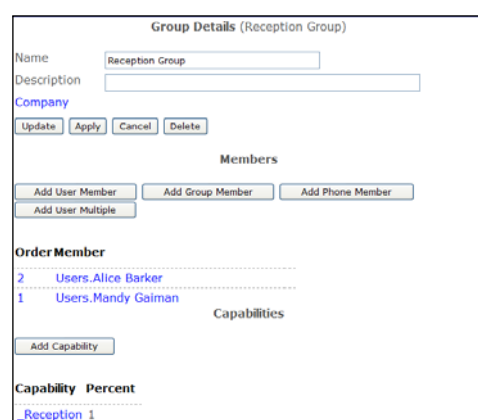
12   Select Update or Apply when ready.

Please note that capabilities are accumulative therefore if a Group requires staff to have 50% fluency in French and 20% fluency in Spanish and you have 10 staff with 50% French of which only 5 have 20% Spanish then the Group will only have 5 members.

A Capability member can be deleted and amended in the same manner as described for a User above.

Enabling Group log in/log out

The Capabilities feature can be used to provide the Users of a PCS 60, PCS 580/570/560 with the ability to log in and log out of a Group.

1 In Manager select Users
2 Select the User required
3 Select the Capability page
4 Select Add Capability
5 In the Capability field enter an underscore followed by the text required, eg _Reception. This will be the text that will be displayed in the Groups screen on the User's phone. This must match the name of the Capability entered in the Group as described below.
6 In the Capability Percent field enter the number that will be matched with the Capability Percentage in the Group, eg 1. This value must be the same or higher than the value entered in the Group's Capability Percent field as described below. This will also determine the User's order in the Group

7 Select Update when ready
8 Repeat steps 2 to 7 for all members of the Group who wish to use this facility.
9 In Manager select Groups
10 Select the Group required
11 Select Add Capability
12 Within the Capability field enter an underscore followed by the text required, eg _Reception. This must match the name of the Capability entered in the Users' configuration form as described above.
13 Within the Capability Percent field enter the number required, eg 1. This value must be the same or lower than the value entered in the Users' Capability Percent field described above.
14 Select Update when ready
15 The Group will be automatically populated with the Users whose value in their Capability Percent field is the same or higher than the value in the Group's Percent field.


16 On a PCS 580 the Groups option will now appear within the Group Membership screen which can be accessed by selecting the Settings icon once.

On a PCS 570/560 the Groups option will now appear in the User's Settings screen. On selecting this option the screen will display the Groups that the User can log in or out of. When the User is logged into a Group the Group will be displayed with a tick, when the User is logged out of a Group the Group will be displayed without a tick. (Please refer to the relevant User Manual for further details.)

On a PCS 60 the Groups that the User can log in or out of will be displayed within the Login dialogue box which is accessed via the Login icon. The User should add a tick to log in to that Group or remove the tick to log out of the Group. (Please refer to the PCS 60 User Manual for further details.)
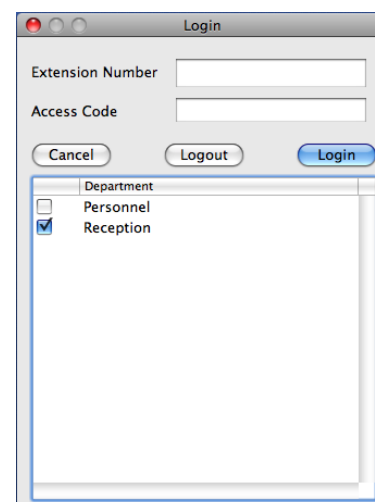
17   When a User logs out of a Group a minus sign is added to their Capability Percent field, eg -1 so that this value no longer matches the Capability requirement of the Group and the User is removed from the Group.
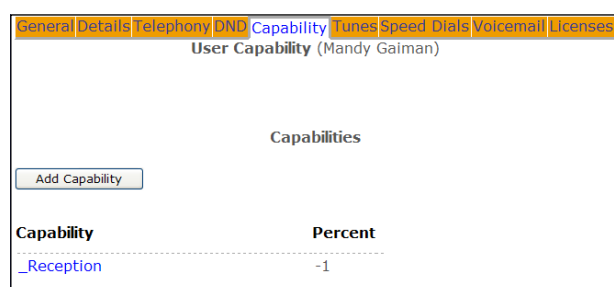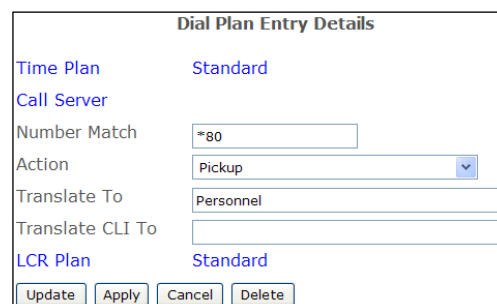

18   When a User logs back into a Group the minus sign is removed and the User is returned to the Group.

Pick up a call to a member of a Group via a Dial Plan entry

A Dial Plan entry similar to this example can be created to allow Users to pick up the longest waiting call ringing for a User that is a member of a Group. This will pick up any internal call, DDI or Department call to a member of the Group.

The name of the Group is entered in the Translate To field.

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	*80
Action	Pickup
Translate To	Personnel
Translate CLI To	
LCR Plan	Standard

Update Apply Cancel Delete

Routing calls via a Department

A Department is used to determine how calls will be presented to a Group. By default, a Department called Main containing MainGroup as its Distribution Group is created. All incoming calls are routed to this Department until configured otherwise.

Creating a Department

Once a Group has been created a Department is then created to determine how calls will be presented to the Group.

1   In Manager select Departments
2   A list of the current Departments are displayed
3   Click on the Add button
4   In the Name field enter a unique name for the Department, use alphanumeric characters only, begin with a letter and do not use any punctuation. This Name is used to identify a call to this Department in Caller Display.
5   If required, in the Description field enter text that will help to identify the Department
6   In the Telephone Number field enter a unique extension number that will be used to route calls to this Department. Ensure that this number is not the same as an existing extension number.
7   Select Update when ready.
8   The new Department will be listed in the Departments List

General	Distribution	Telephony	Voicemail	Licenses

Department General (Sales)

Name	Sales
Description	
Telephone Number	8001

Each Department created will be listed in the Departments directory on a PCS 580/570/560, 410/400, 100, 60 and 50.

Each Department is identified on the database by a GUID (Global Unique Identifier). This unique identifier can be displayed by selecting Utilities and then Voicemail GUIDs.
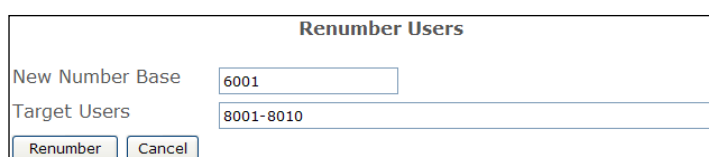
Duplicate Department Names and Extension Numbers

It is essential that duplicate Department Names and Extension Numbers are not used on a system. To ensure that duplicates have not been entered select Utilities and then Sorted Lists. This program will highlight entries where an identical name or extension number exists.

Changing the extension numbers for multiple Departments

When the extension number for more than one Department is to be changed the Users (Renumber) feature can be used as follows:

1   In Manager, select Users (Renumber)
2   In the New Number Base field enter the first number of the new range to be used, for example, if you wish to renumber the extensions from 6001 to 6050 enter 6001.
3   In the Target Users field enter the extension numbers to be changed. Enter either a comma between each extension number or use a hyphen to indicate a range of extensions. For example: 8001,8003,8009,8015 etc or 8001-8010

4   Select Renumber
5   The extension numbers for the selected Departments will be changed, for example, 8001 will become 6001, and the next extension will become 6002 and so on.

Setting Departmental functionality for multiple Departments
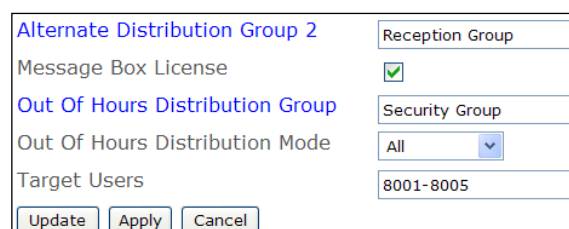
You may wish to turn on a particular feature or multiple features for more than one Department, rather than turn on these feature in each Department's configuration form you can perform this task quickly via the Dept (Bulk) feature in Manager.

1   In Manager, select Dept (Bulk)
2   Select the feature(s) you wish to configure.
3   Select the Continue button at the bottom of the page

4   Select the option required for the relevant field from the list box or tick box provided, or when a field name is displayed in blue the entry for this field can be selected from a list. Click on the field name and select the entry required.
5   In the Target Users field enter the extension numbers to be changed. Enter either a comma between each extension number or use a hyphen to indicate a range of extensions. For example: 8001,8003,8009,8015 etc or 8001-8010
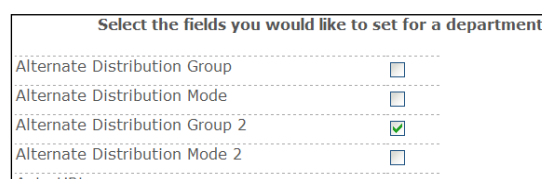

6   Select Update when ready. The Target Departments will be updated with the changes selected.

Assigning a Distribution Group to a Department

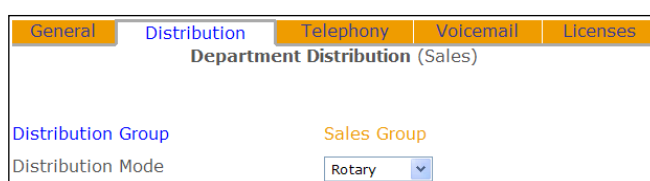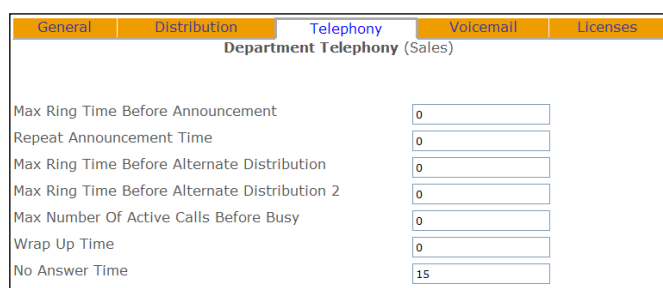A Group is assigned to a Department to determine which Users will receive calls routed to this Department. This is called the Distribution Group. The Group must have been previously created as detailed on page 122.

1 In Manager select Departments
2 A list of the current Departments are displayed
3 Select the Department required
4 Select the Distribution page
5 Select the Distribution Group field
6 From the Select Group list select the Group required
7 This Group will be displayed in the Distribution Group field
8 From the Distribution Mode list box select the method to be used to distribute the calls:

General	Distribution	Telephony	Voicemail	Licenses

Department Distribution (Sales)

Distribution Group   Sales Group

Distribution Mode   Rotary

    a   All – all members of the Group are presented with the call simultaneously. This mode will allow you to reject a call (using a PCS) without it coming back a few seconds later.
    b   Rotary – the call will be presented to the member of the Group with the highest Order number, then to the User with the next highest number and so on.
    c   Sequential – same as Rotary however subsequent calls will be presented to the next User in the Group after the User who was last presented with a call.
    d   Manual – this option will allow third party call centre applications to control when calls will be presented to an agent.

9 Select Apply when ready
10 If Rotary or Sequential Mode is selected the No Answer Time will determine how long the call will ring on each extension before moving to the next extension. Select the Telephony page and in the No Answer Time field enter the number of seconds required.
11 Select Update or Apply when ready.

General	Distribution	Telephony	Voicemail	Licenses

Department Telephony (Sales)

Max Ring Time Before Announcement	0
Repeat Announcement Time	0
Max Ring Time Before Alternate Distribution	0
Max Ring Time Before Alternate Distribution 2	0
Max Number Of Active Calls Before Busy	0
Wrap Up Time	0
No Answer Time	15

Please note:
- Comfort announcements can be played to the caller while waiting for the call to be answered. Please refer to page 216 for further information.
- If the Max Number of Active Calls before Busy entry has been exceeded the caller will be given busy or presented to voicemail.
- Calls will pass to voicemail after the time entered in the Max No Answer Time before Voicemail field. Please refer to page 209 for further details.

Using an Alternate Distribution Group

If members of the Distribution Group are unavailable to take a call it can be passed to an Alternate Distribution Group. The Max Ring Time Before Alternate Distribution field determines how long the call will ring round the Distribution Group without being answered before being passed to the Alternate Distribution Group. The Alternate Distribution Mode will determine how the calls will be presented to the Alternate Distribution Group.

1   In Manager select Departments
2   A list of the current Departments are displayed
3   Select the Department required
4   Select the Distribution page
5   Select the Distribution Skip On No Targets check box if a call is to be presented immediately to the Alternate Distribution Group when all members of the Distribution Group are busy.
Only used with the Rotary and Sequential Distribution modes
6   Select the Alternate Distribution Group field.
7   From the Select Group list select the Group required and this Group will be displayed in the Alternate Distribution Group field

General	Distribution	Telephony	Voicemail	Licenses

Department Distribution (Sales)

Distribution Group	Sales Group
Distribution Mode	Rotary
Distribution Skip On No Targets	☐
Alternate Distribution Group	Post Sales Group
Alternate Distribution Mode	All

8   From the Alternate Distribution Mode list box select the method to be used to distribute the calls:
   a   All – all members of the Group are presented with the call simultaneously. This mode will allow you to reject a call (using a PCS) without it coming back a few seconds later.
   b   Rotary – the call will be presented to the member of the Group with the highest Order number, then to the User with the next highest number and so on.
   c   Sequential – same as Rotary however subsequent calls will presented to the next User in the Group after the User who was last presented with a call.
   d   Manual – this option will allow third party call centre applications to control when calls will be presented to an agent.
9   Select Apply
10   Select the Telephony page, in the Max Ring before Alternate Distribution field enter the number of seconds to elapse before the unanswered call is presented to the Alternate Distribution Group.

General	Distribution	Telephony	Voicemail	Licenses

Department Telephony (Sales)

Max Ring Time Before Announcement	0
Repeat Announcement Time	0
Max Ring Time Before Alternate Distribution	60
Max Ring Time Before Alternate Distribution 2	0
Max Number Of Active Calls Before Busy	0
Wrap Up Time	0
No Answer Time	15

Please note:
- If the Distribution Group Mode is set to All this will be the amount of time the call will ring on all extensions simultaneously before being passed to the Alternate Distribution Group.
- If the Distribution Group Mode is set to Rotary or Sequential this will be the amount of time the call will ring round the Group before being passed to the Alternate Distribution Group. Therefore if the Group contains two members, the No Answer Time is set to 10 seconds and the Max Ring before Alternate Distribution field is set to 30 seconds, the call will ring on the first extension for 10 seconds, on the second for 10 seconds, back to the first for 10 seconds and then be passed to the Alternate Distribution Group.
- If the Distribution Skip on No Targets field is enabled this field will be ignored when all members of the Distribution Group are busy.
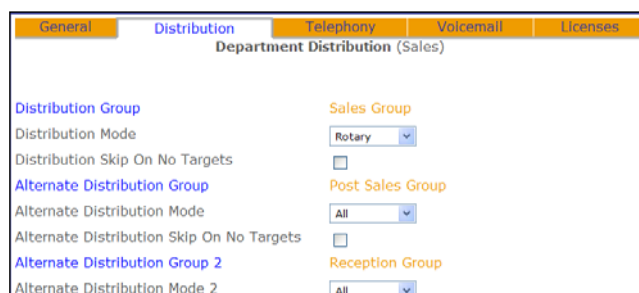11   Select Update or Apply when ready

Please note:
- Comfort announcements can be played to the caller while waiting for the call to be answered. Please refer to page 216 for further information.
- If the Max Number of Active Calls before Busy entry has been exceeded the caller will be given busy.

- Calls will pass to voicemail after the time entered in the Max No Answer Time before voicemail field. Please refer to the Voicemail for a Department section from page 209 for further details.

Using an Alternate Distribution Group 2

If the members of the Alternative Distribution Group are unavailable to take a call it can then pass to the Alternate Distribution Group 2. The Max Ring Time Before Alternate Distribution 2 field determines how long the call will ring round the Alternate Distribution Group without being answered before being passed to the Alternate Distribution Group 2. The Alternate Distribution Mode 2 will determine how the calls will be presented to the Alternate Distribution Group 2.

1   In Manager select Departments
2   A list of the current Departments are displayed
3   Select the Department required
4   Select the Distribution page
5   Select the Alternate Distribution Skip On No Targets check box if a call is to be presented immediately to the Alternate Distribution Group 2 when all members of the Alternate Distribution Group are busy. Only used with the Rotary and Sequential Alternate Distribution modes.
6   Select Alternate Distribution Group 2 field
7   From the Select Group list select the Group required and this Group will be displayed in the Alternate Distribution Group 2 field
8   From the Alternate Distribution Mode 2 list box select the method to be used to distribute the calls:

General	Distribution	Telephony	Voicemail	Licenses

Department Distribution (Sales)

Distribution Group	Sales Group
Distribution Mode	Rotary
Distribution Skip On No Targets	☐
Alternate Distribution Group	Post Sales Group
Alternate Distribution Mode	All
Alternate Distribution Skip On No Targets	☐
Alternate Distribution Group 2	Reception Group
Alternate Distribution Mode 2	All

   a   All – all members of the Group are presented with the call simultaneously. This mode will allow you to reject a call (using a PCS) without it coming back a few seconds later.
   b   Rotary – the call will be presented to the member of the Group with the highest Order number, then to the User with the next highest number and so on.
   c   Sequential – same as Rotary however subsequent calls will presented to the next User in the Group after the User who was last presented with a call.
   d   Manual – this option will allow third party call centre applications to control when calls will be presented to an agent.
9   Select the Telephony page, in the Max Ring Time before Alternate Distribution 2 field enter the number of seconds to elapse before the unanswered call is presented to the Alternate Distribution Group 2.

General	Distribution	Telephony	Voicemail	Licenses

Department Telephony (Sales)

Max Ring Time Before Announcement	0
Repeat Announcement Time	0
Max Ring Time Before Alternate Distribution	60
Max Ring Time Before Alternate Distribution 2	60
Max Number Of Active Calls Before Busy	0
Wrap Up Time	0
No Answer Time	15

   Please note:
   - If the Alternate Distribution Group Mode is set to All this will be the amount of time the call will ring on all extensions simultaneously before being passed to the Alternate Distribution Group 2.
   - If the Alternate Distribution Group Mode is set to Rotary or Sequential this will be the amount of time the call will ring round the Group before being passed to the Alternate Distribution Group 2. Therefore if the Group contains two members, the No Answer Time is set to 10

seconds and the Max Ring before Alternate Distribution 2 field is set to 30 seconds, the call will ring on the first extension for 10 seconds, on the second for 10 seconds, back to the first for 10 seconds and then be passed to the Alternate Distribution Group 2.

- If the Alternate Distribution Skip on No Targets field is enabled this field will be ignored when all members of the Distribution Group are busy.

10   Select Update or Apply when ready

Please note:
- Comfort announcements can be played to the caller while waiting for the call to be answered. Please refer to page 216 for further information.
- If the Max Number of Active Calls before Busy entry has been exceeded the caller will be given busy.
- Calls will pass to voicemail after the time entered in the Max No Answer Time before voicemail field. Please refer to page 209 for further details.

Using Night Service

The Out of Hours Mode field determines the operational state of a Department. By default this is set to Not Used, which means the Out of Hours facilities are not used and calls are always presented to the Distributions Groups as described above.

However this field can be used when it is necessary to route calls to another Group or play a message, for example, when the office is closed, when staff are not at their desk, or when staff are in a meeting, etc

When Out of Hours is selected calls to the Department can be presented to an Out of Hours Distribution Group or passed directly to voicemail. When the office is re-opened, staff are at their desks and available to take calls the In Hours option can then be selected and calls will be presented to the Distribution Groups again.

Both of these Out of Hours modes can be selected manually in Manager, as follows:

1   In Manager, select Departments.
2   Select the Department required
3   From the Out of Hours Mode list box select either Out of Hours or In Hours
4   Select Update or Apply when ready

Alternatively, Dial Plan entries can be used to set the Out of Hours Mode field. This allows staff to set the Out of Hours Mode field themselves, for example, when staff go into a meeting, when the office closes etc.

1   Create two Dial Plan entries similar to the following examples:

Dial Plan Entry Details		Dial Plan Entry Details	
Time Plan	Standard	Time Plan	Standard
Call Server		Call Server	
Number Match	*90	Number Match	*91
Action	Night Service Out Of Hours	Action	Night Service In Hours
Translate To	8001	Translate To	8001
Translate CLI To		Translate CLI To	
LCR Plan	Standard	LCR Plan	Standard
Update Apply Cancel Delete		Update Apply Cancel Delete	

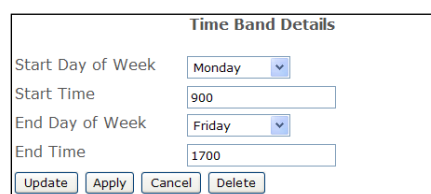(Please refer to the Working with Dial Plans section from page 114 for further information.)

2   On dialling, eg *90, the Out of Hours Mode field will be set to Out of Hours.
3   On dialling, eg *91, the Out of Hours Mode field will be set to In Hours.

Using a Time Plan

The Timed Out of Hours Mode is selected when an Time Plan is used to specify the exact hours the Department is In Hours, for example, when the office hours are 9.00 to 5.30 or the Accounts department is always closed between 12.30 and 1.30, etc. Within the hours specified by the Time Plan calls will be passed to the Distribution Groups. Outside of the hours specified by the Time Plan calls can be passed to an Out of Hours Distribution Group or directly to voicemail.

Firstly, a Time Plan must be created as follows:

1   In Manager select Time Plans
2   Click on the Add button
3   Enter a Name using alphanumeric characters only and enter a Description, if required
4   Select Apply
5   Select Add Time Band
6   In the Start Day of Week list box select the day of the week the Time Plan is to start
7   In the Start Time field enter the time each day the Time Plan is to start
8   In the End Day of Week list box select the day of the week the Time Plan is to end
9   In the End Time field enter the time each day the Time Plan is to end
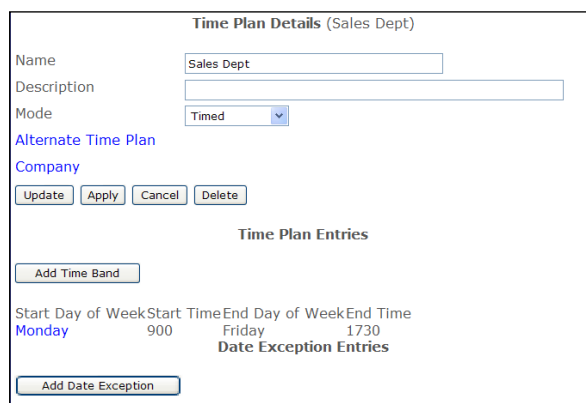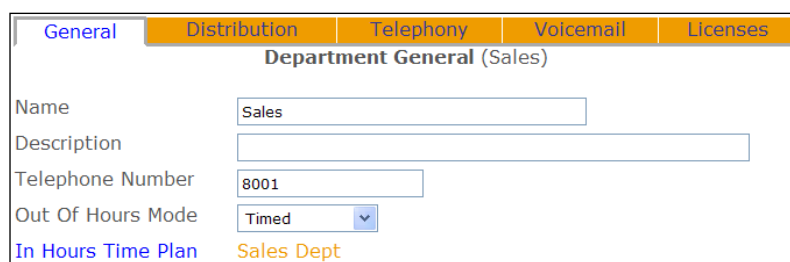10   Select Update
11   Select Update or Apply when ready.

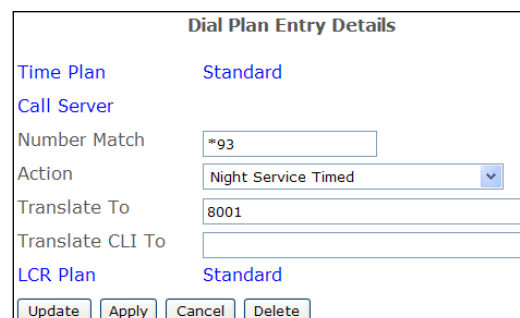(Please refer to Using a Time Plan section from page 148 for further information.)

The Time Plan created above is then entered into the Department's In Hours Time Plan field, as follows:

1   Select Departments.
2   Select the Department required
3   From the Out of Hours Mode list box select Timed
4   Select Apply
5   Select the In Hours Time Plan field and from the Select Time Plan list select the Time Plan created above
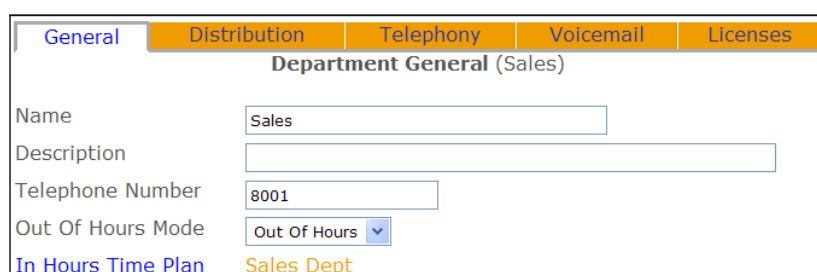
6   Select Update or Apply when ready.

A Dial Plan entry can also be used to place a Department into Timed mode as per this example

Viewing a Department's Out of Hours Mode

The Out of Hours Mode field within a Department's configuration form will display the mode currently being used by the Department.

1   In Manager select Departments.
2   Select the Department required
3   The Out of Hours Mode field will display Not Used, In Hours, Out of Hours or Timed.

View the status of a Department

The current status of a Department (either In Hours or Out of Hours) can also be viewed by opening:

http://<ip address of call server>/manager/time.php
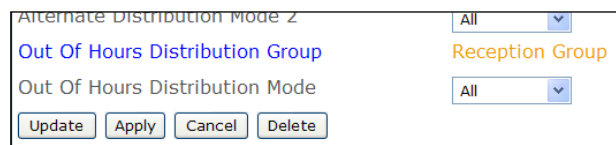
eg http://192.168.0.1/manager/time.php

Using an Out of Hours Distribution Group

When a Department's Out of Hours Mode field is set to Out of Hours calls can be passed to an Out of Hours Distribution Group. When a Department's Out of Hours Mode field is set to Timed calls can be

passed to an Out of Hours Distribution Group when the time is outside of the Time Bands specified in the Time Plan.

An Out of Hours Distribution Group is configured as follows:

1   In Manager select Departments.
2   Select the Department required
3   Select the Distribution page
4   Select the Out of Hours Distribution Group field
5   From the Select Group list select the Group required and this will be displayed in the Out of Hours Distribution Group field
6   From the Out of Hours Distribution Mode list box select the method to be used to distribute the calls to the Out of Hours Distribution Group:
   a   All – all members of the Group are presented with the call simultaneously. (This mode will allow you to reject a call (using a PCS) without it coming back a few seconds later.)
   b   Rotary – the call will be presented to the member of the Group with the highest Order number, then to the User with the next highest number and so on.
   c   Sequential – same as Rotary however subsequent calls will presented to the User with the next highest number in the Group after the User who was last presented with a call.
   d   Manual – this option will allow third party call centre applications to control when calls will be presented to an agent.
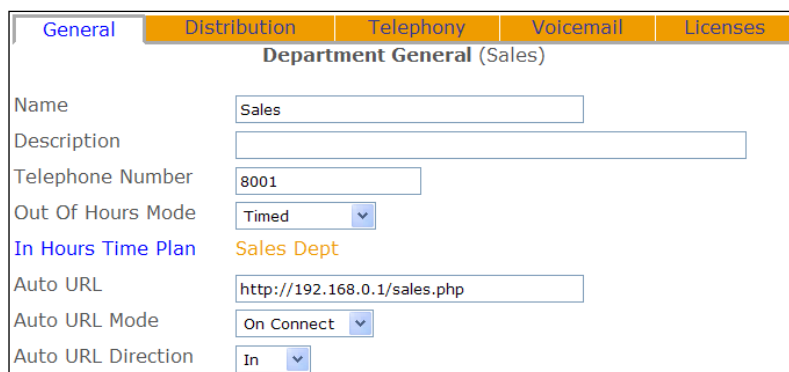7   Select Update or Apply when ready.

Please note:
•   Comfort announcements can be played to the caller while waiting for the call to be answered. Please refer to the Voicemail for a Department section from page 209 for further information.
•   If the Max Number of Active Calls before Busy entry has been exceeded the caller will be given busy.
•   Calls will pass to voicemail after the time entered in the Out of Hours Max No Answer Time before voicemail field. Please refer to the Voicemail for a Department section from page 209 for further information.

Using an Auto URL

If a member of a Department is using a PCS 60, 50 or 410/400 a web page can be displayed when a call is made to and/or received for a Department. This could be a page stored on the internal web server (please refer to the Internal Web Server section from page 184 for further details), the customer's intranet or accessed externally if an Internet connection is available.

1   In Manager select Departments
2   From the Departments list select the Department required.
3   In the Auto URL field enter the path to the web page to be displayed, eg www.splicecom.com or http://192.168.0.1/sales.php
4   From the Auto URL Mode list box select one of the following:
   a   NoAction – the web page is not automatically displayed on a PCS when a call is made to and/or received for the Department. However the web page can be viewed via the Information icon on PCS.
   b   On Ring – the web page will display on the PCS when a call for this department is ringing (this is useful if a script is to be read when answering a call).
   c   On Connect – the web page will display on the PCS once a call to this department has been answered.

5   From the Auto URL Direction list box select one of the following:
    a   Both – the web page will be displayed on the PCS making the call and on the PCS receiving the call for this Department
    b   In – the web page will only be displayed on the PCS receiving the call for this Department
    c   Out – the web page will only be displayed on the PCS making the call to this Department

6   Select Update or Apply when ready.

General	Distribution	Telephony	Voicemail	Licenses

Department General (Sales)

Name	Sales
Description	
Telephone Number	8001
Out Of Hours Mode	Timed
In Hours Time Plan	Sales Dept
Auto URL	http://192.168.0.1/sales.php
Auto URL Mode	On Connect
Auto URL Direction	In

(Please note that the Auto URL settings for a Contact take priority over the Auto URL settings for a Department.)

Displaying a Directory

If a member of a Department is using a PCS 50 or 410/400 a list of Users or Departments or the User's speed dials can be displayed when a call is made to and/or received for a Department to help them process the call efficiently. This feature would be useful to an Operator/Receptionist who will be transferring calls to other Users or Departments.

1   In Manager select Departments
2   From the Departments list select the Department required.
3   From the Auto URL Mode list box select one of the following:
    a   Favourites – the User's speed dial list will be displayed once a call to this department has been answered
    b   Users – the User directory will be displayed once a call to this department has been answered
    c   Departments – the Department directory will be displayed once a call to this department has been answered
4   From the Auto URL Direction list box select one of the following:
    a   Bothway – the directory will be displayed on the PCS making the call and on the PCS receiving the call for this Department
    b   Incoming – the directory will only be displayed on the PCS receiving the call for this Department
    c   Outgoing – the directory will only be displayed on the PCS making the call to this Department
5   Select Update or Apply when ready.

(Please note that the Auto URL settings for a Contact take priority over the Auto URL settings for a Department.)

Setting the maximum number of calls to be handled by the Department

The Max Number of Active Calls before Busy feature will determine the total number of calls to be controlled by the Department at any one time. This number includes all connected calls, all calls waiting and all calls being distributed, including calls being handled by voicemail. Once this number has been exceeded any subsequent caller will be given busy.

1 In Manager select Departments
2 From the Departments list select the Department required.
3 Select the Telephony page
4 In the Max Number of Active Calls before Busy field enter the number of calls to be handled by the Department at any one time.
5 Select Update or Apply when ready.

Assigning a Wrap Up Time

The Wrap Up Time feature specifies the amount of time given to each member of the Distribution Groups at the end of each call. By default, this is set to 2 seconds and if configured must set to 2 or greater. They will not be available to receive the next call until this time has elapsed. This will allow time for any administrative tasks eg completing contact reports, updating databases etc. This time will affect any User who has answered a call for this Department regardless of how the connection was made.

1 In Manager select Departments
2 From the Departments list select the Department required.
3 Select the Telephony page
4 In the Wrap Up Time field enter the number (2 or greater) of seconds to elapse before a User will be presented with the next call.

5   Select Update or Apply when ready.

General	Distribution	Telephony	Voicemail	Licenses

Department Telephony (Sales)

Max Ring Time Before Announcement	0
Repeat Announcement Time	0
Max Ring Time Before Alternate Distribution	60
Max Ring Time Before Alternate Distribution 2	60
Max Number Of Active Calls Before Busy	10
Wrap Up Time	120

The Wrap Up Time can be truncated:

- by user invention ie by placing a phone off-hook and then on-hook or by making a call.
- on a PCS 580/570/560 by selecting the Wrap Up key.
- on a PCS 100 by selecting the Exit key.

Using the Wrap Up Time on a PCS 60, PCS 50 and PCS 410/400

If a PCS 60, PCS 50 or PCS 410/400 is to receive calls for a Department that has a Wrap Up Time set the Auto URL field must contain an address to a web page, eg www.splicecom.com, that will be displayed during the Wrap Up Time. The Auto URL Mode field can be set to No Action. For further information please refer to the Using an Auto URL section from page 136. The User of a PCS 60 can truncate the Wrap Up Time by selecting the Hang Up icon. The User of a PCS 50 or PCS 410/400 can truncate the Wrap Up Time by selecting the Exit icon.

Using Call Waiting

Call Waiting will allow members of the Distribution Group to be aware when calls to the Department are waiting in the queue while they are on a call.   When this feature is enabled the relevant Users will be informed that a call to the Department is waiting to be answered by an intermittent beep in the headset and, if relevant, via their PCS 60, 50, 410/400 or 580/570/560. The User can then toggle between their current call and the next call in the queue.

In order to use this facility the relevant Users must also have Call Waiting set in their User configuration form as described on page 91.

Configuring Call Waiting

1   In Manager select Departments
2   Select the Department you wish to configure
3   Select the Telephony page
4   From the Call Waiting list box select one of the following options:
    a   Disable (default) – the Call Waiting feature is not set for this Department.
    b   Enable – Users will be informed that a call is waiting in the queue by an intermittent beep in the headset and via their PCS 580/570/560, 410/400, 100, 60 or 50. The User can receive a second call and toggle between the two calls.
    c   Phone Tools – if the User is running a PCS 60 or 50 as a partner to their handset this option will enable Call Waiting and the intermittent beep will not be heard in the headset but can be heard via the PCS 60/50. If the User is not using a PCS 60/50 as a partner this option is the same as setting the Enable option.

d   NoBeep – enables Call Waiting but disables the intermittent beep. The User will be informed of a call waiting on the screen of their PCS 580/570/560, PCS 410/400, PCS 100, PCS 60 or PCS 50.

5   Select Update or Apply when ready

Call Waiting to an analogue handset can be handled as follows:

1   First call is in progress and the User is informed of a call waiting by the intermittent beep
2   Use the Hold facility to place the first call on hold
3   The second call will be automatically presented
4   Use the Hold facility to end the second call and return to the first caller.
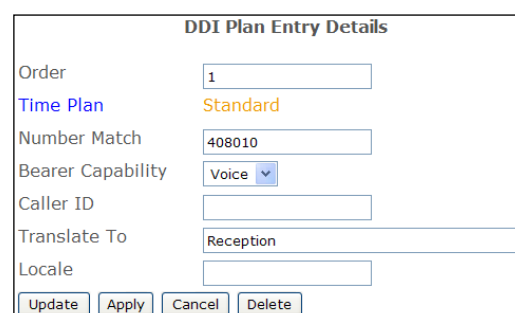
If the User replaces the handset instead of following Step 4 the two calls will be connected together.

(Please refer to the relevant User Manual for details on how to handle a call waiting via a PCS 50, 60, 100, 410/400 or 580/570/560.)

Using a DDI Plan to Route calls to the Department

External calls to a Department can be routed via a DDI entry. (Please refer to the Working with DDI Plans from page 169 for further details.)

1   In Manager select DDI Plans
2   Select the DDI Plan required
3   Select Add DDI Entry
4   Select the new DDI Entry
5   In the Number Match field enter the DDI number to be used
6   From the Bearer Capability list box select Voice
7   In the Translate To field enter the extension number or name of the Department
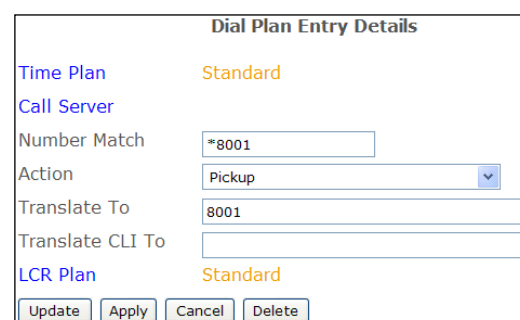8   Select Update or Apply when ready.

DDI Plan Entry Details

Order	1
Time Plan	Standard
Number Match	408010
Bearer Capability	Voice
Caller ID	
Translate To	Reception
Locale	

Update Apply Cancel Delete

Pick up a Department call via a Dial Plan entry

A Dial Plan entry can be created similar to this example that will allow Users to pick up the longest waiting call ringing for a Department.

The Extension number or Department Name can be used in the Translate To field.

Dial Plan Entry Details

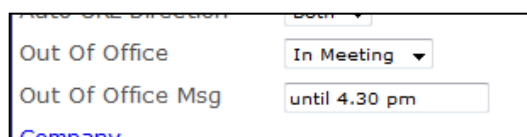|	
Time Plan	Standard
Call Server	
Number Match	*8001
Action	Pickup
Translate To	8001
Translate CLI To	
LCR Plan	Standard

Update Apply Cancel Delete

Using an Out of Office Message

When all members of a Department are out of the office, in a meeting for example, they may wish to inform internal callers of their whereabouts. An Out of Office message can be configured, such as In meeting until 4.30 pm, and this will be displayed on the caller's PCS 50, 60, 100, 410/400, 560/570/580.

This feature can be set up as follows:

1	In Manager select Departments
2	From the Departments list select the Department required
3	Select the General page
4	From the Out of Office list box select the mode required, eg In Meeting.
5	In the Out of Office Message field enter the text to appear with the above selection, eg until 4.30 pm.

Out Of Office	In Meeting ▼
Out Of Office Msg	until 4.30 pm
Company	

6	Select Update or Apply when ready
7	The message will appear on a caller's PCS 50, 60, 100, 410/400, 560/570/580 when they make a call to this Department (please refer to relevant user manual for further details.)

When an Out of Office message has been selected and a caller goes to voicemail the relevant voicemail greeting will be played, please refer to the Recording an Out of Office greeting from page 219 for further details.

To cancel an Out of Office message select "In Office" from the Out of Office list box. The text in the Out of Office Message field can remain to be used next time this feature is applied.

Voicemail

For information on using voicemail functionality with a Department please refer to page 209.

Music on Hold

For information on assigning music on hold to a specific Department please refer to page 179.

Setting up the Contacts Database

The Contacts database can be used to store information on, for example, customers and suppliers. The database can be used as an address book, a telephone directory for speed dialling and for matching the incoming CLI. The Contacts database can store up to 10,000 entries.

Creating a Contact

1	In Manager select Contacts
2	Select the Add button
3	Enter a unique Name that will be used to identify this Contact. Use alpha-numeric characters only and do not use any punctuation.
4	Complete the remaining fields as required, for example
	a	Enter the contact's telephone numbers that will be used either for dialling or for matching with incoming CLI
	b	Enter the contact's company name and department

c   Within the Organisation page enter the contact's job title, address etc
5   Select Update or Apply when ready.

	General	Organisation	Notes

Contact General (Mark Oak)

Name	Mark Oak
Description	
Company	Ocean Electronics
Department	Sales
Direct Route To	
Telephone Number	01727234567
Mobile Telephone Number	07890138388
Fax Telephone Number	
Home Telephone Number	
Spare 1 Telephone Number	
Spare 2 Telephone Number	
Ambiguous Telephone Number	
Email Address	
Auto URL	
Auto URL Mode	No Action
Auto URL Direction	Both
Ex Directory	☐

Update Apply Cancel Delete

Each Contact created will be listed in the Contacts Directory on a PCS 60, 50, 100, 410/400 and 580/570/560. The phone numbers entered can be used by the PCS for speed dialling. When a call is received if the incoming CLI is matched to a Contact the Name and Company fields will be displayed on the PCS.   An analogue handset with caller display will display the Name. The address details of the Contact entered within the Organisation page can also be viewed via a PCS 60, 50 and 410/400.

Please note that the Company field in a Contact bears no relation to the Companies feature of the system. This is purely the name of the business in which the Contact works. For information on creating Contacts for use with the Companies feature please refer to page 155

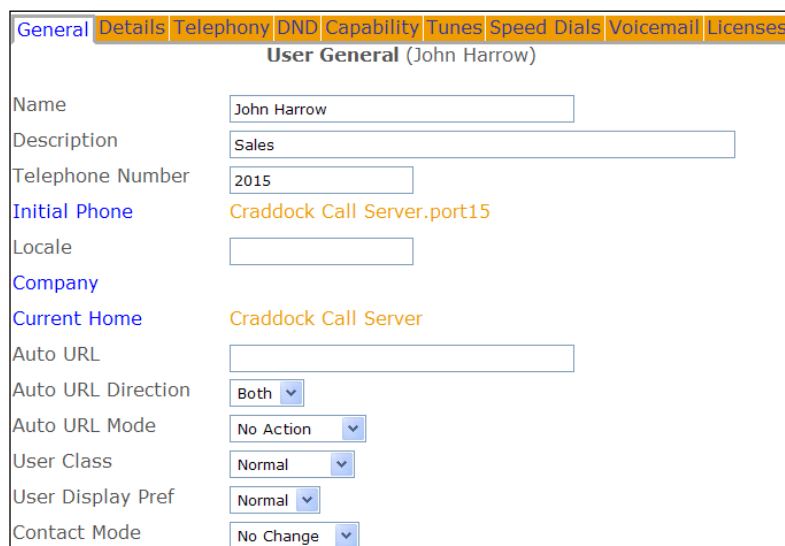Please refer to page 241 for details on how to import or export a Contacts database.

Setting a User's rights to the Contact database

By default all Users have full access to the Contacts database and if they are users of a PCS 60, 50 or 410/400 they can create and amend Contact entries. This functionality is determined by a User's Contact Mode field. This option can be used to reduce a User's rights to the Contacts database so that they cannot make changes to the database by creating a new entry or amending existing entries. They will still be able to view the information and use the numbers for speed dialling.

A User's rights to the Contacts database can be configured as follows:

1   In Manager select Users
2   Select the User required
3   Within the General page from the Contact Mode list box select one of the following:

a   Add/Update – this is the default entry. The User will be able to create new entries and amend existing entries

b   Update – the User will be only able to amend existing entries.

c   No Change – the User will not be able to create new entries or amend existing entries.

4   Select Update or Apply when ready.

PCS 50 and PCS 410/400

When the Update option is selected the New Contact icon will not be available, however the Information icon will be displayed when a Contact is selected and changes to the Contact can be made.

When the No Change option is selected the New Contact icon will not be available, the Information icon will be displayed when a Contact is selected but any changes made will not be saved.

PCS 60

When the Update option is selected the Add Contact icon will not be active. The Information icon will be available to view the Contact details and any changes can be saved.

When the No Change option is selected the Add Contact icon will not be active. The Information icon will be available to view the Contact details but the Save/OK button will be unavailable and any changes will not be saved.

Using an Ambiguous Number

The Ambiguous Number field can be used to match a range of numbers, particularly useful if a company has a large DDI number range. The ? is used to specify any number, therefore an entry of, eg 01923 2877??, will match all calls within the range of 01923 287700 to 287799. Contacts with a complete match will have priority over an ambiguous number.

A Contact with an ambiguous number can be created as follows:

1   In Manager select Contacts
2   Select the Add button
3   Enter a unique Name that will be used to identify this Contact. Use alpha-numeric characters only and do not use any punctuation.
4   In the Ambiguous Number field enter the number to be matched against incoming CLI, using a ? where a digit will be variable, eg 01923 287???
5   Select Update or Apply when ready.

General	Organisation	Notes

Contact General (Ocean Electronics)

Name	Ocean Electronics
Description	
Company	
Department	
Direct Route To	
Telephone Number	01727234000
Mobile Telephone Number	
Fax Telephone Number	
Home Telephone Number	
Spare 1 Telephone Number	
Spare 2 Telephone Number	
Ambiguous Telephone Number	01727234???

Using an Auto URL

When a call is received from or made to a Contact it may be necessary to present the User with a specific web page on their PCS 60, 50 or 410/400 to help them deal with the call efficiently. This could be a page stored on the internal web server (please refer to page 184 for further details), the customer's intranet or accessed externally if an Internet connection is available.

1   In Manager select Contacts
2   From the Contact list select the Contact required.
3   In the Auto URL field enter the path to the web page to be displayed, eg www.splicecom.com or http://192.168.0.1/sales.php
4   From the Auto URL Mode list box select one of the following:
   a   NoAction – the web page specified in the Auto URL field is not automatically displayed on a PCS when a call is made to and/or received from this Contact. However the web page can be viewed via the Information icon on the PCS.
   b   On Ring – the web page will display on the PCS when a call from or to this Contact is ringing (this is useful if a script should be read when answering a call).
   c   On Connect – the web page will display on the PCS once a call from or to this Contact is answered.
5   From the AutoURL Direction list box select one of the following:
   a   Bothway – the web page will be displayed on a PCS when a call is made to this Contact and on a PCS when a call is received from this Contact
   b   In – the web page will only be displayed on a PCS when a call is received from this Contact
   c   Out – the web page will only be displayed on a PCS when a call is made to this Contact
6   Select Update or Apply when ready.

(Please note that the Auto URL settings for a Contact take priority over the Auto URL settings for a Department.)

Displaying a Directory

When a call is received from or made to a Contact a list of Users or Departments or the User's speed dials can be displayed on the PCS 50 or 410/400 answering the call to enable the call to be processed efficiently.

1 In Manager select Contacts
2 From the Contacts list select the Contact required.
3 From the Auto URL Mode list box select one of the following:
    a Favourites – the User's speed dial list will be displayed once a call to or from this Contact has been answered
    b Users – the User directory will be displayed once a call to or from this Contact has been answered
    c Departments – the Department directory will be displayed once a call to or from this Contact has been answered
4 From the AutoURL Direction list box select one of the following:
    a Bothway – the directory will be displayed on a PCS when a call is made to this Contact and on a PCS when a call is received from this Contact
    b In – the directory will only be displayed on a PCS when a call is received from this Contact
    c Out – the directory will only be displayed on a PCS when a call is made to this Contact
5 Select Update or Apply when ready.

Contact Notes

During a call to or from a Contact, when using a PCS 60, 50 or 410/400, a note can be written and stored with the Contact for future information. The User must select Add to Contact if the note is to be stored permanently on the database.

These notes can be viewed within the Notes page of the Contact. Each note can contain up to 128 characters.

For further information on using this feature please refer to the PCS 60 or PCS 410/400 and PCS 50 User Manuals.

Routing calls from a Contact to a specific User or Department

Calls from a particular Contact may need to be handled by a specific User or Department. The **max**imiser can route calls directly rather than via the receptionist first, for example. This feature can be configured as follows:

Firstly, determine to whom the Contact will be directed:

1   Open the relevant Contact
2   In the Direct Route To field enter the extension number, User or Department Name to which a call is to be routed.
3   Select Update or Apply when ready.

General	Organisation	Notes

Contact General (Ellington Telecoms)

Name	Ellington Telecoms
Description	
Company	
Department	
Direct Route To	8001

Secondly, all the relevant DDI Plan entries containing the telephone numbers that the Contact may use must be amended to contain square brackets followed by the extension number, User or Department Name to which a call will be normally routed, as per this example.

Please refer to the Working with DDI Plans section on page 169 for further information.

DDI Plan Entry Details

Order	1
Time Plan	Standard
Number Match	408010
Bearer Capability	Voice
Caller ID	
Translate To	[]Reception
Locale	

Update   Apply   Cancel   Delete

If the Contact belongs to a specific Company please refer to page 158 for details on how to ensure the system checks the correct Contacts database.

Generating a Nuisance Number

The Contacts database can also be used to blacklist specific incoming numbers from which users of a PCS 60, 50 and 410/400 do not wish to receive calls. This is achieved by creating a Contact called ZZZNUISANCE as follows:

1   Create a new Contact.
2   In the Name field enter *ZZZNUISANCE*
3   In the Direct Route To field enter one of the following:
    a   Extension number – this could be the extension number of a User or Department that will filter the calls, eg Reception, or
    b   *!Play:wav file name*, eg !Play:mymessage.wav – this could play a recording to the caller indicating that their call is not required. The WAV file must be stored in the FTP area of the Call Server. For information on how to place a WAV file in the FTP area please refer to the Configuring Music Channels section on page 177.

General	Organisation	Notes

Contact General (ZZZNUISANCE)

Name	ZZZNUISANCE
Description	
Company	
Department	
Direct Route To	8010

After receiving a nuisance call the users of a PCS 60, 50 and 410/400 can now blacklist this external number as follows:

- PCS 410/400 and PCS 50 - by accessing the Incoming Call List, selecting the call and clicking on the Nuisance Number icon .

- PCS 60 – by accessing the Message screen, selecting the call and clicking on the Blacklist icon .

Please refer to the relevant User Manual for further details.

A new Contact will be created in the Contacts database where the Contact Name will be *ZZZNUISANCEexternal number* eg ZZZNUISANCE01923283838. The entry in the Direct Route To field will also be copied into this new Contact

In order for the Direct Route To feature to operate all the relevant DDI Plan entries containing the telephone numbers that the nuisance caller might ring must be amended as described above in the Routing calls from a Contact to specific User or Department section.

Using a Time Plan

A Time Plan can be used to specify the day of the week and time of the day a facility is operational. For example, a Time Plan can be used to stipulate when a Department is operational and outside of these hours when the Out of Hours Distribution Group will be used.
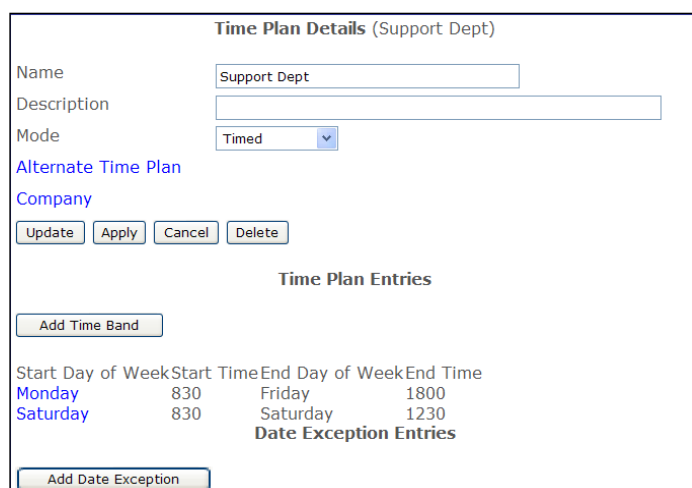
A Time Plan can be applied to Departments, Dial Plans, LCR Plans, DDI Plans and VXML Scripts.

By default, a Time Plan called Standard is created. This is set for Saturday to Sunday, 00:00 to 23:59 so that all features using a Time Plan will be active 24 hours unless changed. It is highly recommended that this Time Plan is not amended.

Creating a Time Plan

1 In Manager select Time Plans
2 Click on the Add button
3 Enter a Name, using alphanumeric characters only, begin with a letter and do not use any punctuation, and enter a Description if required
4 In the TimeZone field enter the time zone required for this Time Plan, eg GMT, UTC, Eire, etc
5 Note that the Mode field is set to Timed by default.
6 Select Apply
7 Select Add Time Band
8 In the Start Day of Week list box select the day the Time Plan is to start
9 In the Start Time field enter the time each day the Time Plan is to start, in 24 hour format eg 0930
10 In the End Day of Week list box select the day the Time Plan is to end

11   In the End Time field enter the time each day the Time Plan is to end, in 24 hour format eg 1745
12   Select Update
13   Repeat steps 6 to 11 to create further entries
14   Select Update or Apply when ready.
15   This Time Plan is now available to be entered in any Time Plan field of a relevant manageable object eg a Department

Please note:

- A Time Band will not be operational if the time crosses midnight on Saturday eg Saturday 1800 Monday 800. This example should be entered as two separate time bands eg Saturday 1800 Saturday 2359, Sunday 0 Monday 800.
- A Time Plan determines when a facility is available, eg a Department is In Hours, and outside of these hours when a facility is not available, eg a Department is in Out of Hours. Therefore you do not need to create a second Time Plan that covers the hours outside of the assigned Time Plan.
- When a Time Plan is used with an internet connection this plan is only checked at initial connection. Therefore a Time Plan will not disconnect the connection.

For information on the time provided by the system please refer to page 30.

Overriding a Time Plan

A Time Plan has three modes – Timed, Out of Hours and In Hours. By default each Time Plan is set to Timed mode and the Time Plan will use the Time Band(s) configured. However it is possible to override the Time Bands by the use of the Mode field. When the In Hours mode is selected the Time Plan is operational, in other words it considers itself to be inside the hours specified by the Time Band(s) regardless of the day or time. When Out of Hours mode is selected the Time Plan considers itself to be outside of the hours specified by the Time Band(s) regardless of the day or time. This means that a facility, such as a Dial Plan entry or VXML Script, configured with a Time Plan can be placed in In Hours or Out of Hours, or, in other words, enabled or disabled, whenever required.

These three Modes can be selected manually in Manager, as follows:

1   In Manager, select Time Plans.
2   Select the Time Plan required
3   From the Mode list box select either Timed, Out of Hours or In Hours
4   Select Update or Apply when ready

Alternatively, a Dial Plan entry can be used to set the Mode field similar to the following examples:

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	*81
Action	Time Plan Enable
Translate To	Sales Dept
Translate CLI To	
LCR Plan	Standard

Update Apply Cancel Delete

This Dial Plan entry would set the Mode field to In Hours. The Translate To field contains the name of the Time Plan to be configured.

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	*82
Action	Time Plan Disable
Translate To	Sales Dept
Translate CLI To	
LCR Plan	Standard

Update Apply Cancel Delete

This Dial Plan entry would set the Mode field to Out of Hours. The Translate To field contains the name of the Time Plan to be configured.

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	*83
Action	Time Plan Timed
Translate To	Sales Dept
Translate CLI To	
LCR Plan	Standard

Update Apply Cancel Delete

This Dial Plan entry would set the Mode field to Timed. The Translate To field contains the name of the Time Plan to be configured.

Using an Alternate Time Plan

On certain days of the year, eg bank holidays, an alternative Time Plan may be required because, for example, the phones are only going to be manned for the morning or the company is closed and calls are to be routed directly to voicemail, etc

1   Firstly, determine the operational hours to be used on, for example, a bank holiday. If there are no operational hours when, for example, the office is closed on a bank holiday leave the Alternative Time Plan field blank.   If, for example, phones are going to manned in the morning on a bank holiday, create a new Time Plan configured with the alternative Time Bands similar to the following example:

2   Return to the original Time Plan
3   Select the Alternative Time Plan field
4   From the Select Time Plan list select the Time Plan required, eg Bank Holidays
5   Select Apply

Secondly, you will need to enter the dates that the Alternative Time Plan is to be used as follows:

1   Select Add Date Exception
2   In the Day of Month field enter the date required
3   From the Month list box select the month required
4   Select Update
5   Repeat steps 6 to 9 until the all dates required have been entered
6   Select Update or Apply when ready.

View the status of a Time Plan

The current status of a Time Plan (either In Hours or Out of Hours) can be viewed by opening:

http://<ip address of call server>/manager/time.php

eg http://192.168.0.1/manager/time.php
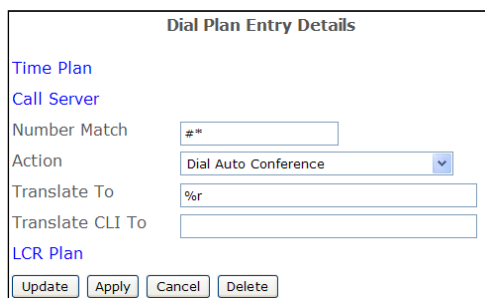
Conferencing

The Conferencing feature allows Users to create a 3-way conversation between themselves and two internal and/or external calls via their handset or create a Meet Me Conference whereby a conference is created and Users and external callers then join this conference.

To use this facility the Conference Server option must be ticked within the relevant Voicemail Port. Each participant of a conference occupies a voicemail connection, therefore a Voice Processing Port licence will be required for each connection. Please refer to the Voicemail Ports section from page 188 for further information.

3-way Conferencing

A User can create a 3-way conference call via their PCS 60/50 or PCS 580/570/560/410/400/100. Please refer to relevant User manual for instructions on how to create a 3-way conference using these handsets.

Alternatively a 3-way conference can be created via a Dial Plan entry similar to the following examples:
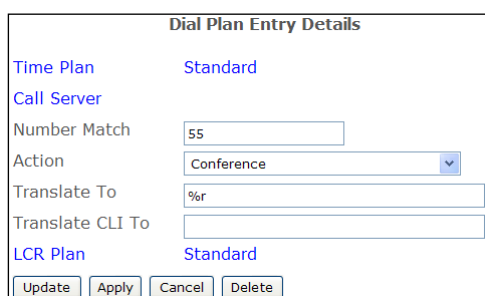
Dial Plan Entry Details

Time Plan
Call Server
Number Match   #*
Action   Dial Auto Conference
Translate To   %r
Translate CLI To
LCR Plan
[Update] [Apply] [Cancel] [Delete]

This default Dial Plan entry will allow a User to join a call and create a conference. The number (internal or external) to be called when creating the conference is entered in the Translate To field. (Commonly known as "barge")
For example, a call is active between extensions 2001 and 2002, extension 2003 can dial either #*2001 or #*2002 to join the conversation.

Dial Plan Entry Details

Time Plan   Standard
Call Server
Number Match   55
Action   Conference
Translate To   %r
Translate CLI To
LCR Plan   Standard
[Update] [Apply] [Cancel] [Delete]

This example Dial Plan entry will allow a User to create a conference with two internal or external calls. The number (internal or external) to be called when creating the conference is entered in the Translate To field.
For example, extension 2001 makes a call to extension 2002 and puts the call on hold. Extension 2001 would then dial 552003. Once the call has been answered extension 2001 would press Recall and the three are in a conference.

Setting up a Meet Me Conference
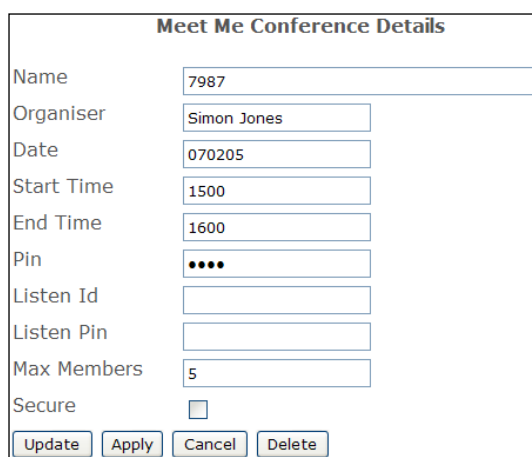
The Meet Me Conference facility can be used to create a call conference that callers can join, via a DDI Plan entry or Dial Plan entry, whilst the conference is in progress.

A Meet Me Conference can be created either via Manager, DDI Plan entry or via a Dial Plan entry.

Creating a Meet Me Conference

1   In Manager select Meet Me Conferences
2   Select the Add button
3   Enter a number in the Name field that will be used to identify the conference

4   If you wish to identify the organiser of this conference enter the text/numbers required in the Organiser field

5   If users are to be limited to a specific date and time that they can join the conference enter the following:

a   In the Date field enter the date (in yymmdd format) on which the conference will take place.

b   In the Start Time field enter the time from which users can join the conference, eg 1100, with no punctuation.

c   In the End Time field enter the time you wish the conference joining to end, eg 1200, with no punctuation.

6   If the ability to initiate and join the conference is to be protected by a password, in the Pin field enter a number which users must enter when starting or joining the conference.

7   If the conference is to be limited to a specific number of users, in the Maximum Members field enter the maximum number of participants for this conference.

8   Select Update or Apply when ready.
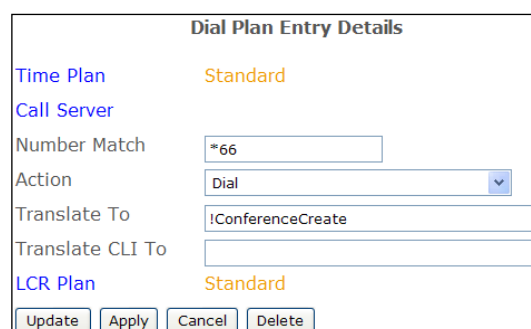
The conference is then initiated via a Dial Plan or DDI Call Plan entry as follows.

Initiating a Meet Me Conference

A Meet Me Conference can be started via a Dial Plan entry similar to this example.

After dialling the Dial Plan entry, eg *66, the user will be asked to enter the conference number followed by #, eg 7987#. If a number has been entered in the Pin field for this conference the user will be asked to enter the conference PIN followed by #, eg 1234#. A beep will confirm that the conference has been successfully started.

The following alternatives could be entered in the Translate To field:

1   !ConferenceCreate:conference number , eg !ConferenceCreate:7987.

2   !ConferenceCreate:%r
The User would dial the short code followed by the conference number and #, eg *997987#.

In both cases the User would be asked for the conference PIN if configured and a beep would confirm that the conference has been successfully started.

A Meet Me Conference can be started via a DDI Plan entry similar to this example.

After dialling the DDI number the user will be asked to enter the conference number followed by #, eg 7987#. If a number has been entered in the Pin field for this conference the user will be asked to enter the conference PIN followed by #, eg 1234#. A beep will confirm that the conference has been successfully started.

DDI Plan Entry Details

Order	1
Time Plan	Standard
Number Match	407987
Bearer Capability	Voice
Caller ID	
Translate To	!ConferenceCreate
Locale	

Update Apply Cancel Delete

Users and external contacts can then join a conference either via a Dial Plan entry or DDI Plan entry as described below in Joining a Meet Me Conference.

Both the Dial Plan entry and DDI Plan entry described above can be used to create and start a new Meet Me Conference. When asked for the conference number any number can be entered unless used for an existing Meet Me Conference. A Meet Me Conference entry will be automatically created and can be viewed via Manager. The extension number or DDI number of the originator of the conference will be displayed in the Organiser field. Users can join this conference either via a Dial Plan entry or DDI Plan entry as described below within 30 minutes of its creation. If the User belongs to a Company the Meet Me Conference entry will be displayed under the specific Company within Manager. Please refer to the Working with Companies section from page 155 for further information.
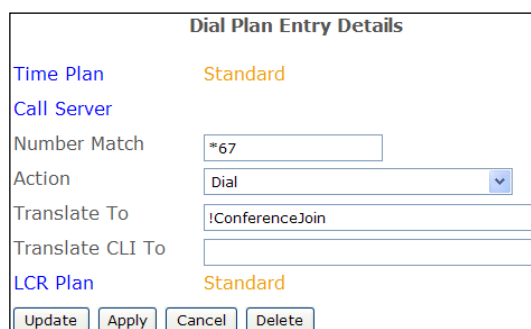
A Dial Plan entry can also be used to create and start a Meet Me Conference where the conference number is the extension number of the User that originated the conference. In this case the Translate To field would contain *!ConferenceCreate:%s*.

Joining a Meet Me Conference

A Meet Me Conference can be joined via a Dial Plan entry or DDI Plan entry.

To allow Users to join a Meet Me Conference internally a Dial Plan similar to this example can be used.

When a User dials this Dial Plan entry, eg *67, they will firstly be asked for the conference number of the Meet Me Conference they wish to join followed by #, eg 7987#. If a number has been entered in the Pin field for this conference the user will be asked to enter the conference PIN followed by #, eg 1234#. A beep will confirm that they have successfully joined the conference.

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	*67
Action	Dial
Translate To	!ConferenceJoin
Translate CLI To	
LCR Plan	Standard

Update Apply Cancel Delete

The following alternatives could be entered in the Translate To field:
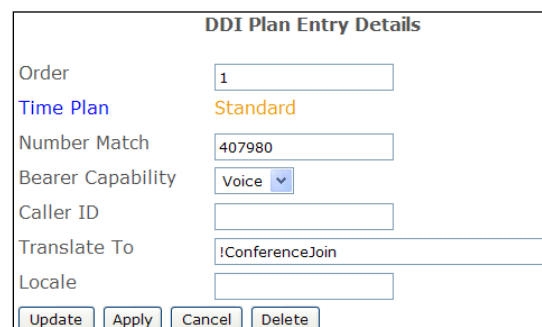
1   !ConferenceJoin:conference number , eg !ConferenceJoin:7987.

2   !ConferenceJoin:%r

The User would dial the short code followed by the conference number and a #, eg *677987#.

In both cases the User would be asked for the conference PIN if configured and a beep would confirm that they have successfully joined the conference.

A DDI Plan entry can be used to allow external staff or contacts to join a Meet Me Conference, as per this example.

When a caller dials this DDI number they will firstly be asked for the conference number of the Meet Me Conference they wish to join followed by #, eg 7987#. If a number has been entered in the Pin field for this conference the user will be asked to enter the conference PIN followed by #, eg 1234#.   A beep will confirm that they have successfully joined the conference.
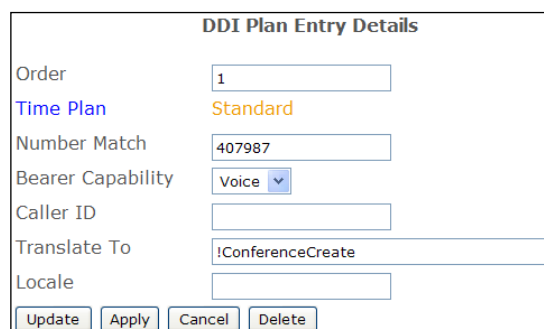
DDI Plan Entry Details

Order	1
Time Plan	Standard
Number Match	407980
Bearer Capability	Voice
Caller ID	
Translate To	!ConferenceJoin
Locale	

[Update] [Apply] [Cancel] [Delete]

Working with Companies

The Companies feature allows the **max**imiser system to be used by multiple businesses/ companies. For example, where a building is occupied by more than one business and one **max**imiser system is servicing the whole building.

The Companies feature ensures that these businesses:
- only view their own User, Department and Contact Directories on a PCS,
- only access their own Park slots, and
- only display their own ringing phones in the Call Pick Up screen on the PCS 50 and 410/400.
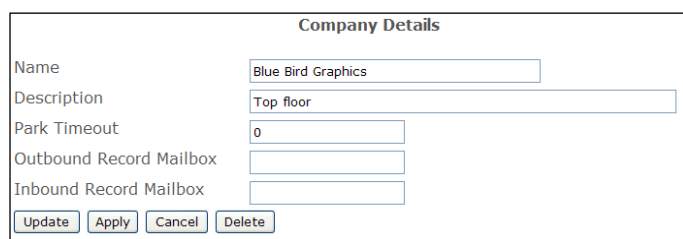
Each Company can also have its own auto attendant and Park Timeout.

The database is still centrally administered via Manager providing ease of management of the system however users will be unaware of this usage.

For information on recording calls to a specific Company please refer to the Call Recording section on page 232.

Creating a Company

1   In Manager select Companies
2   A list of the current Companies are displayed
3   Select the Add button
4   Enter a Name to identify the Company. Use alphanumeric characters only, begin with a letter and do not use any punctuation.
5   If required, in the Description field enter a description to help to distinguish the Company
6   If required, in the Park Timeout field enter the amount of time (in seconds) a call parked by a User who is a member of this Company is held in the park slot before being represented to the User. By default, this is set to 5 minutes.
7   Select Update or Apply when ready.

Company Details

Name	Blue Bird Graphics
Description	Top floor
Park Timeout	0
Outbound Record Mailbox	
Inbound Record Mailbox	

[Update] [Apply] [Cancel] [Delete]

Each Company is identified on the database by a GUID (Global Unique Identifier). This unique identifier can be displayed by selecting Utilities and then Voicemail GUIDs.
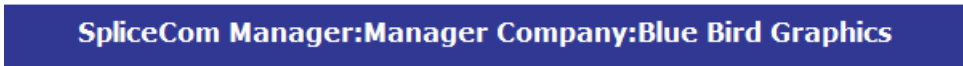
For further information on using a Park Timeout and parking a call please refer to the Using the Park Facility section from page 106.

For information on how to record inbound and outbound external calls made by a member of a Company please refer to the Call Recording section from page 232.

Managing a Company

Each Company can have its own Contacts database, Meet Me Conferences and Auto Attendant; and specific Users, Departments, Groups, Time Plans and Assistants can be assigned to the Company. These can be administered via the Manage Company feature as follows.

1   In Manager select Manage Company
2   From the Select Company to Manage list select the Company required
3   The Company that you are currently managing will be displayed in the title bar

SpliceCom Manager:Manager Company:Blue Bird Graphics

4   Select either Users, Departments, Contacts or Meet Me Conferences to manage these objects within the Company

Users

Users created while managing a specific company will be automatically assigned to that Company.

Users assigned to a Company will only be able to view, via the Users Directory on their PCS, Users assigned to the same company. The User will still be able to dial other Users on the system however they will not be able to do this via the Users Directory.

Users assigned to a Company can use the Call Pick-up feature on their PCS 50 and 410/400 for extensions who are also members of the same company. The relevant extension must be set up in their Speed Dial list.

Park slots are company dependent; therefore a User will not be able to pick up a parked call parked by a User belonging to different Company or a global User (a User not assigned to a specific Company). Please refer to the Using the Park Facility section from page 106 for further information.

Departments

Departments created while managing a specific company will be automatically assigned to that Company.

A User belonging to this Company will now only be able to view, in the Departments Directory on their PCS, Departments assigned to the same company. The User will still be able to dial other Departments on the system however they will not be able to do this via the Departments Directory.

Groups

Groups created while managing a specific company will be automatically assigned to that Company.

Assigning a Group to a Company will enable an Administrator who has been assigned to the same Company to view and administer the Groups within Manager.

Time Plans

Time Plans created while managing a specific company will be automatically assigned to that Company.

Assigning a Time Plan to a Company will enable an Administrator who has been assigned to the same Company to view and administer the Time Plans within Manager.

Contacts

Contacts created while managing a specific company will be solely for the use of this Company.

Only Users of a PCS that belong to the same Company will be able to view these Contacts.

Global Users (User who have not been assigned to a Company) will only be able to view Contacts created in the global Contacts list.

A Contact can be created by a User via their PCS 60, 50 or 410/400; if this User belongs to a Company the Contact will be saved in the relevant Company's Contact list. A Contact created by a global User will be saved in the global Contacts list.

Incoming CLI will only be matched to entries within the Company to which the User receiving the call belongs.

Meet Me Conferences

Meet Me Conferences created while managing a specific company will be solely for the use of this Company. Please refer to page 152 for further details on working with a Meet Me conference.

The conference can be initiated via a Dial Plan or DDI Call Plan entry where the Translate To field would contain *!ConferenceCreate:company name:*, eg !ConferenceCreate:Blue Bird Graphics:

Users and external contacts can join the conference via a Dial Plan or DDI Call Plan entry where the Translate To field would contain *!ConferenceJoin:company name:*, eg !ConferenceJoin:Blue Bird Graphics:

Auto Attendant

Please refer to page 225 for details on how to create an Auto Attendant for a Company.

Clear the Manage Company feature
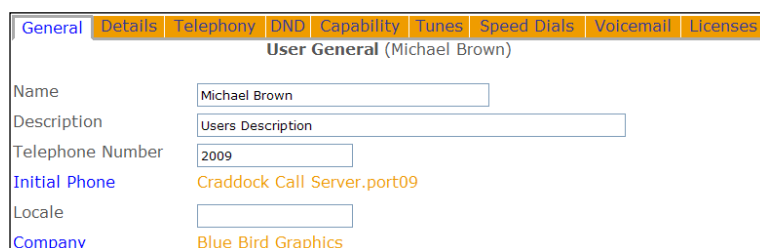
When you wish to return to managing the complete database the following steps can be taken:

1   In Manager select Manage Company
2   Select Clear
3   The title bar will display that the Manage Company feature is no longer on and all Users, Departments etc can be viewed.

Assigning an existing User/Department/Group/Time Plan to a Company

An existing User, Department, Group or Time Plan on the system can be assigned to a Company if required.

1   In Manager select either Users, Departments, Groups, or Time Plans
2   Select the User, Department, Group or Time Plan required
3   Select the Company field
4   From the Select Company list select the Company required and this will be entered into the Company field
5   Select Update or Apply when ready.

General	Details	Telephony	DND	Capability	Tunes	Speed Dials	Voicemail	Licenses

User General (Michael Brown)

Name	Michael Brown
Description	Users Description
Telephone Number	2009
Initial Phone	Craddock Call Server.port09
Locale	
Company	Blue Bird Graphics

A User that is not assigned to a Company will remain a global User and will be able to view all Users regardless of the Company to which they belong. This is useful if, for example, a centralised reception is being used for all Companies.

A global User will be able to view all Departments regardless of the Company they belong to.

Creating a Default Company

All Users, Department, Contacts etc that do not belong to a specific Company are global Users, global Departments and so on. If you wish to configure features available via a Company for the global entries, such as the Park Timeout, call recording etc, create a Company called Default and configure the feature required. Do **not** add the global Users, global Departments etc to this Company.
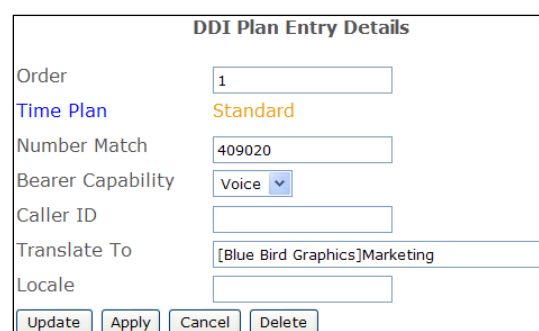
Routing calls from a Companies' Contact to a specific User or Department

Calls from a particular Contact may need to be handled by a specific User or Department, the **max**imiser can route calls directly rather than via the receptionist first, for example. However, if the Contact belongs to a Company the system needs to know which Company's list of Contacts to search. This feature can be configured as follows:

Firstly, determine to whom the Contact will be directed:

1 Open the relevant Contact
2 In the Direct Route To field enter the extension number, User or Department Name to which a call is to be routed.
3 Select Update or Apply when ready.

Secondly, all the relevant DDI Plan entries containing the telephone numbers that the Contact may use must be routed to the required Company so that the system knows which Company's list of Contacts to search. The Translate To field must contain the Company name in square brackets followed by the extension number, User or Department Name to which a call will be normally routed, as per this example.
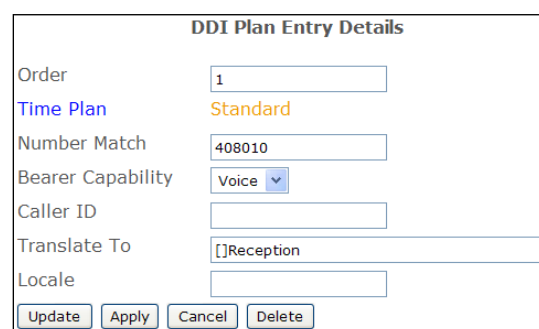
DDI Plan Entry Details

Order	1
Time Plan	Standard
Number Match	409020
Bearer Capability	Voice
Caller ID	
Translate To	[Blue Bird Graphics]Marketing
Locale	

Update Apply Cancel Delete

If the Contact to be routed is contained within the global Contacts list do not enter a Company Name in the square brackets, as per this example.

Please refer to the Working with DDI Plans section from page 169 for further information on creating new DDI Plan entries.
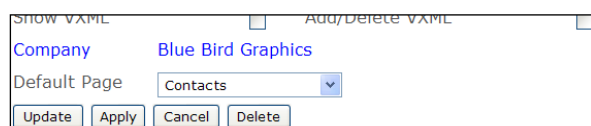
DDI Plan Entry Details

Order	1
Time Plan	Standard
Number Match	408010
Bearer Capability	Voice
Caller ID	
Translate To	[]Reception
Locale	

Update Apply Cancel Delete

Creating an Administrator for a Company

When creating an Administrator to view and manage the database as described from page 31 you may wish this Administrator only to have access to the entries relating to a specific Company. In this case within the required Administrator's configuration form select the Company field and select the required Company from the Select Company list.

Show VXML	☐	Add/Delete VXML	☐
Company	Blue Bird Graphics		
Default Page	Contacts		

Update Apply Cancel Delete
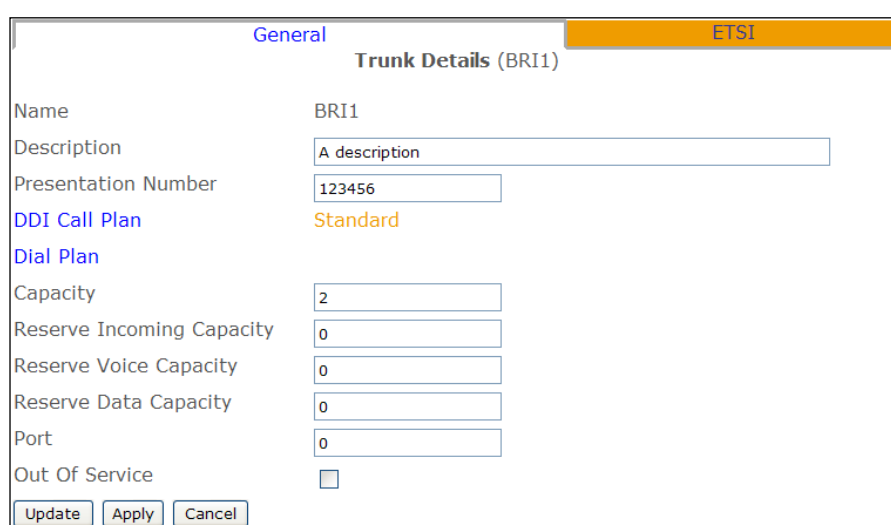
Working with Trunks

Configuring a Trunk

Each ISDN BRI or PRI portprovided by a Call Server or Trunk Module is displayed under the Trunk section of the configuration. Each Trunk can be configured independently, as follows.

1 In Manager select Trunks
2 Select the relevant module containing the trunk required, eg CallServer
3 Select the trunk required

4   You may wish to change the Description field to assist you with identifying the port

5   In the Presentation Number field enter the number to be presented as outgoing CLI on this port if required. (Please note that the Translate CLI To field in a Dial Plan entry and a DDI Plan entry will have priority over the Presentation Number. Please refer to the Outgoing CLI for external calls section from page 171 for further details.)

6   Enter a DDI Plan in the DDI Call Plan field if required, this determines

    a   the plan to be used to route an incoming call received on this Trunk, and

    b   the outgoing CLI to be sent with an outgoing call via this Trunk

7   The Capacity field indicates the number of channels available on this Trunk. Amend this value if additional Trunk licences have been purchased as described below.

8   If required, in the Reserve Incoming Capacity field enter the number of channels to be reserved solely for incoming calls.

9   Select Update or Apply when ready.

General	ETSI

Trunk Details (BRI1)

Name	BRI1
Description	A description
Presentation Number	123456
DDI Call Plan	Standard
Dial Plan	
Capacity	2
Reserve Incoming Capacity	0
Reserve Voice Capacity	0
Reserve Data Capacity	0
Port	0
Out Of Service	☐

[Update] [Apply] [Cancel]

Please note: If a Trunk is not configured with either a DDI Call Plan or a Dial Plan calls on this Trunk will receive busy.

Trunk Licencing

A Trunk licence (one licence per channel) is required to:

- activate the channels on the third and fourth BRI ports on the 5100 Call Server
- activate the channels on the two PRI ports available on the 5100 Call Server
- activate the channels on the second BRI port on the 5108 Call Server
- use more than one channel on a PRI trunk on the 4100 Call Server and 4200 Trunk Module.
- activate the channels on the third and fourth BRI ports on the 4140 Remote Call Server.
- activate the channels on the 15 channel PRI port available on the 4140 Remote Call Server.

The Capacity field within the configuration form for each of these ports will be set with a default value of 0 channels and must be amended to enable the channels.
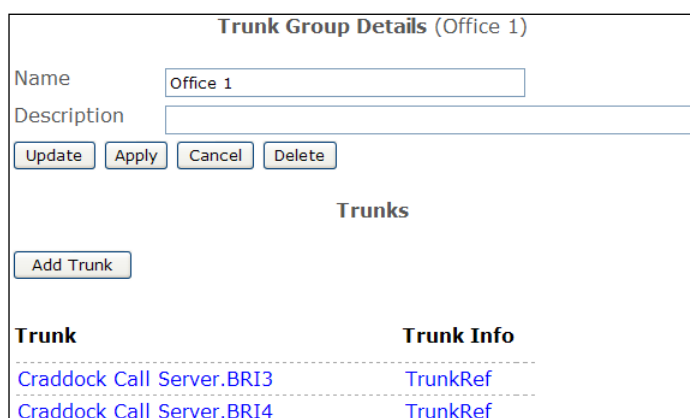
Creating a Trunk Group

Trunk Groups allow one or more Trunks to be grouped together so that outgoing calls can be routed on to specific Trunks. LCR Plan entries point to a Trunk Group so that when specific digits are dialled these calls will be routed to the correct Trunk(s).

By default, all Trunks are members of the AllTrunks group, and the default entry in the Standard LCR Plan points to the AllTrunks group. Therefore all outgoing calls, by default, will use the first available Trunk.

However, further Trunk Groups can be created as follows:

1   In Manager select Trunk Groups
2   Select the Add button
3   Enter a Name, using alpha-numeric characters only, begin with a letter and do not use any punctuation, and a Description if required
4   Select Apply
5   Select Add Trunk
6   Select the Trunk field
7   From the Select Module for Trunks list select the relevant module, eg Call Server
8   From the Select Trunk list select the Trunk required
9   Select Update
10   Repeat steps 5 to 9 to add further Trunks
11   Select Update or Apply when ready.
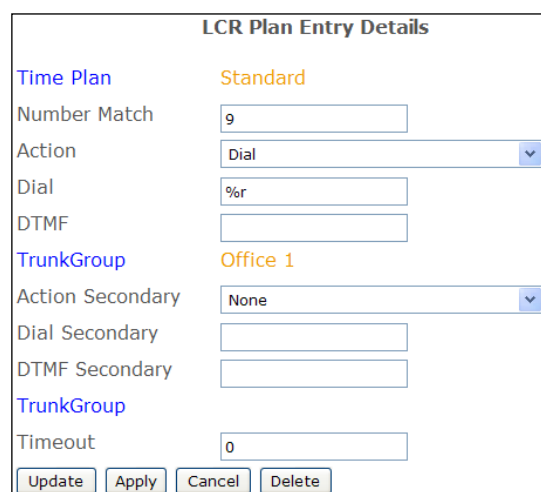
Please note: if a Trunk Group contains trunks from multiple Call Servers, trunks located on the Call Server where the call was initiated will be used first.

Assigning a Trunk Group to a LCR Plan entry

Outgoing calls are routed to a specific Trunk Group via a LCR Plan entry.

1   In Manager select LCR Plans
2   Select the LCR Plan required
3   Select Add LCR Plan Entry

4   In the Number Match field enter the digits to be dialled to access the Trunk Group. (Ensure this entry does not clash with a Dial Plan entry.)

5   From the Action list box select Dial

6   In the Dial field enter the number to be dialled by the system, or
     If all the digits entered by eg the User are to be used enter *%n*, or
     If the digits entered minus the numbers in the Number Match field are to be used enter *%r*

7   Select Apply

8   Select the Trunk Group field

9   From the Select Trunk Group list select the Trunk Group required

10   Select Update when ready.

11   Select Update or Apply when ready.

LCR Plan Entry Details

Time Plan	Standard
Number Match	9
Action	Dial
Dial	%r
DTMF	
TrunkGroup	Office 1
Action Secondary	None
Dial Secondary	
DTMF Secondary	
TrunkGroup	
Timeout	0

Update   Apply   Cancel   Delete

Please refer to the Working with LCR Plans section from page 175 for further information.

Route of an Incoming Call

The following diagram displays the route of an incoming call to the system.

Call is received by a

```
      1
    Trunk
```

Number dialled matched against

```
        2
   DDI Plan Entry
```

Call presented to

```
          3
  User or Department etc
```
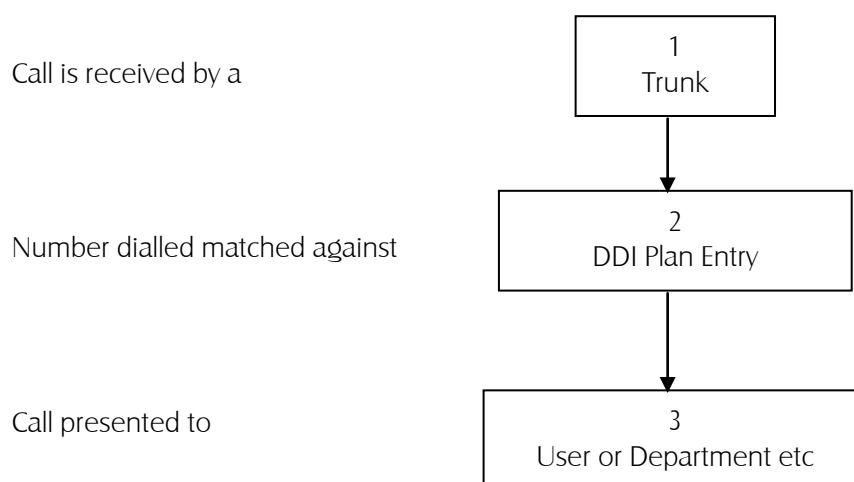
Route of an Outgoing Call

The following diagram displays the route of an outgoing call made by a User of the system.

Number dialled by a	1 User
Number matched to a Dial Plan entry	2 Dial Plan
Number matched to a LCR Plan entry	3 LCR Plan
LCR plan specifies the	4 Trunk Group
Call presented to a (Outgoing CLI determined by DDI Plan or Presentation Number)	5 Trunk
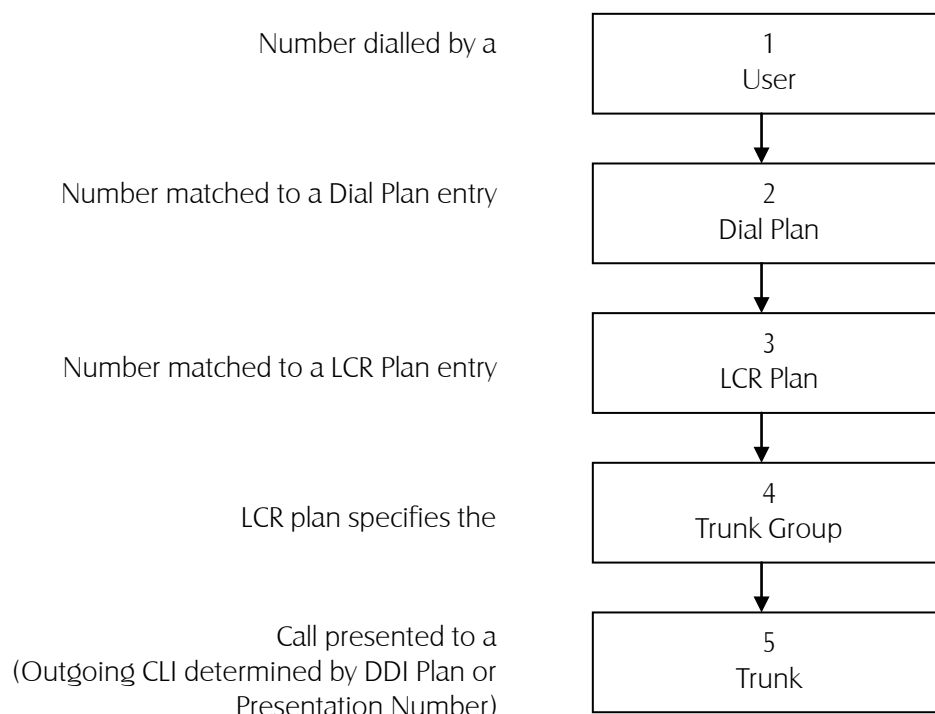
Setting up a Trunk port as a NT port

All ISDN ports on a Call Server or Trunk Module can work in NT mode. Network Termination mode is pretending to be an exchange so that other PBXs can connect to the system.

NT mode can be configured as follows:

1   In Manager select Trunks
2   Select the relevant module containing the trunk required, eg CallServer
3   Select the trunk required
4   Change the Description of the port if required
5   In the Presentation Number field enter the number to be presented as CLI on this port if required. (Please note that the Translate CLI to field in a Dial Plan entry and a DDI Call Plan entry will have priority over the Presentation Number.)
6   Select the ETSI page.
7   In the TEI Number field enter 127 (this may solve communication problems with the remote end. This is not required on a PRI Trunk port).
8   Select the NT Mode field. A tick will indicate that this field is enabled.
9   Select Update or Apply when ready.

To route calls via this Trunk port the following can be configure:

10   From the navigation pane select Trunk Groups
11   Select the Add button to create a new Trunk Group
12   Enter a Name, eg NT Port, and description if required
13   Select Apply
14   Select the Add Trunk button
15   Select the Trunk field
16   From the Select Module for Trunks list select the relevant module
17   From the Select Trunk list select the Trunk modified above
18   Select Update
19   Select Update or Apply when ready.
20   From the navigation pane select LCR Plans
21   Select the relevant LCR Plan eg Standard
22   Select the Add LCR Plan Entry button
23   In the Number Match field enter the digit(s) to be used to access the NT port. (Ensure this entry does not clash with a Dial Plan entry.)
24   From the Action list select Dial
25   In the Dial field enter the number to be dialled by the system, or
      a   If all the digits entered by eg the User are to be used enter *%n*
      b   If the digits enter minus the numbers in the Number Match field are to be used enter *%r*
26   Select Apply
27   Select the Trunk Group field
28   From the Select Trunk Group list select the Trunk Group created above

29   Select Update
30   Select Update or Apply when ready.
31   Connect the relevant dongle to the relevant port and connect this to the device requiring ISDN

Please note:
- No call progress tones are provided
- When setting up Trunk ports to simulating ISDN into your system in a test environment the call can go out on any port other than the NT port.

Trunk NT Mode Dongles
For technical data on NT mode dongles please refer to page 341.

Configuring DPNSS
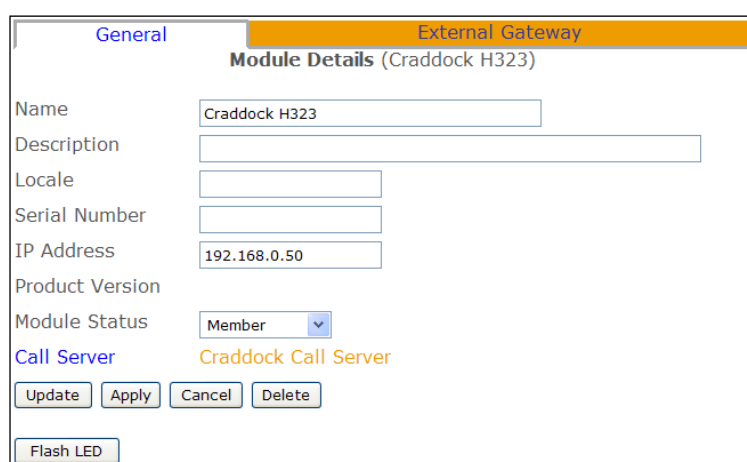
A PRI port provided by a Call Server or Trunk Module can be configured to support DPNSS. No additional licence is required for this feature other than a Trunk licence to enable a channel on the PRI port. This implementation of DPNSS will support: Simple Calls, Name, Number, Loop Avoidance, Data Call, and Node Numbers including Message Waiting and Visual Voice for Index.

Please refer to SpliceCom Support for assistance with configuration of DPNSS.

Creating a H.323 trunk
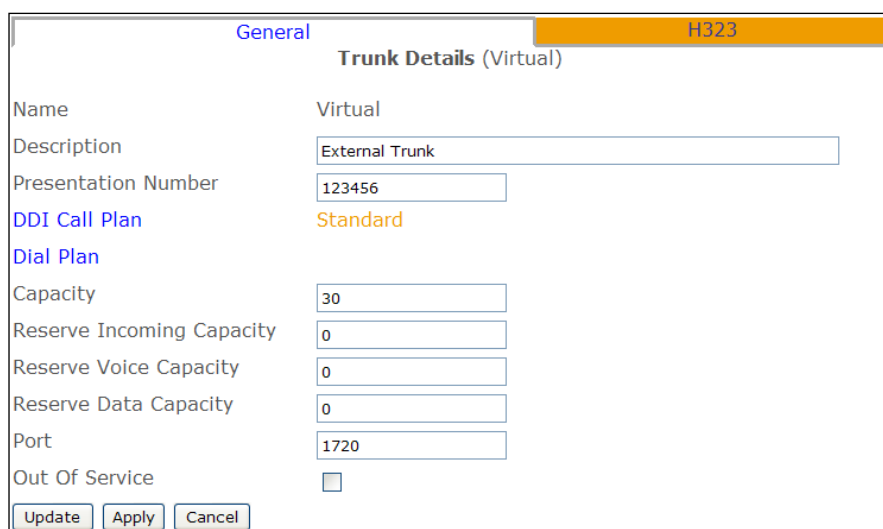
A Call Server can support H.323 Trunk services and can be configured as follows. Please note that the required number of Trunk licences (one per channel) must be entered to use this facility (please refer to page 35 for further details). When using a 4100 Call Server or 4140 Remote Call Server a Voice Compression Module or card is required to use this facility across the Internet (please refer to page 45 for further details).

1. In Manager select Modules
2. From the Module list select the Add H323 Gateway button
3. In the Name field enter a name to identify this external gateway
4. Enter the IP address of the remote end.
5. Select Apply
6. Select the Call Server field
7. From the Select Call Server list select the Call Server to which this H.323 Gateway is to be registered.
8. Select Update or Apply when ready.

9. From the navigation pane select Trunks
10. Select the Module created above

11   From the Trunk List select Virtual
12   In the Presentation Number field enter the number to be sent on this Trunk.
13   Select Apply
14   Select the DDI Call Plan field
15   From the Select DDI Plan list select the DDI Plan required.
16   In the Capacity field enter the number of connections to be allowed on this link.

17   Select the H323 Trunk page
18   From the Compression Type list box select G729A8K, if required, otherwise select None.
19   If the link is going across the Internet, in the NAT Server field enter the IP address allocated by the ADSL provider, otherwise leave this field blank.
20   If the link is going across the Internet, in the NAT Port field enter *1720*, otherwise leave this field blank.
21   If the link is going across the Internet, from the NatMode field select Static, otherwise select None.
22   Select Update or Apply when ready.
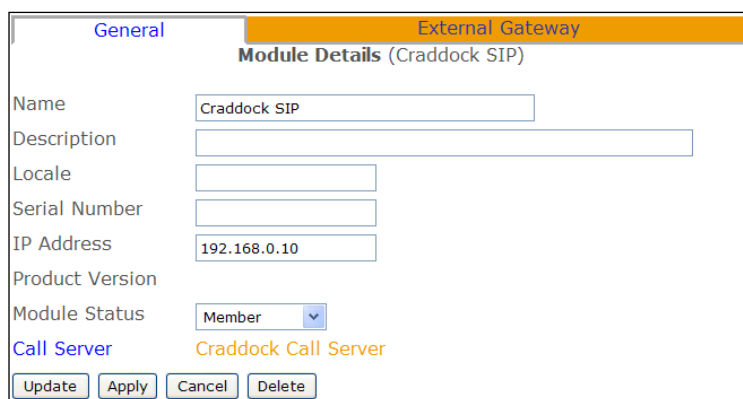
Creating a SIP trunk
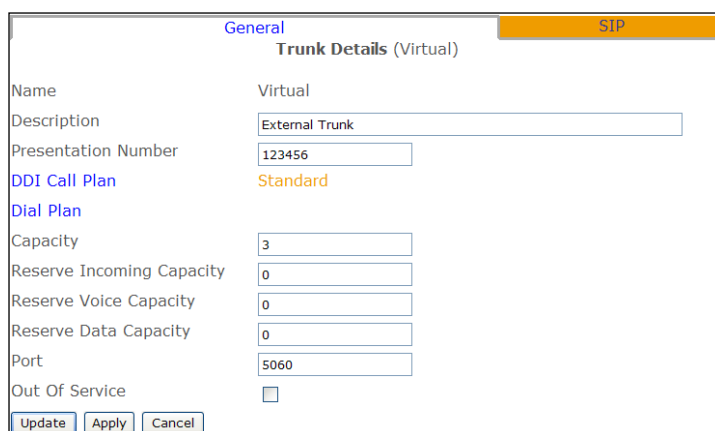
A Call Server can support SIP Proxy Server and SIP Trunk services and can be configured as follows. Please note that when using a 4100 Call Server or 4140 Remote Call Server a Voice Compression Module or card

is required to use this facility (please refer to page 45 for further details) together with the required number of Trunk licences, one per channel (please refer to page 35 for further details).

1   In Manager select Modules
2   From the Module list select the Add SIP Gateway button
3   In the Name field enter a name to identify this external gateway
4   Enter the IP address of the remote end, or the name, e.g. sip.example.com, if required instead
5   Select Apply
6   Select the Call Server field
7   From the Select Call Server list select the Call Server to which this SIP Gateway is to be registered.
8   Select Update or Apply when ready.
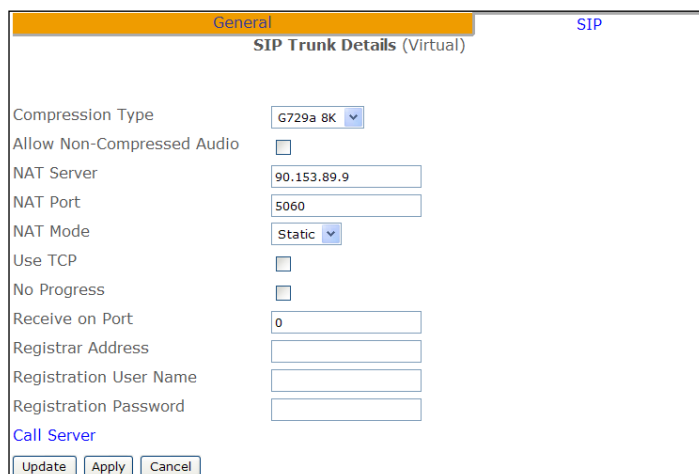
General	External Gateway
Module Details (Craddock SIP)	

Name	Craddock SIP
Description	
Locale	
Serial Number	
IP Address	192.168.0.10
Product Version	
Module Status	Member
Call Server	Craddock Call Server

Update Apply Cancel Delete

9   From the navigation pane select Trunks
10   Select the Module created above
11   From the Trunks List select Virtual
12   In the Presentation Number field enter the number to be sent on this Trunk. (This may be the account number supplied by the SIP provider.)
13   Select Apply
14   Select the DDI Call Plan field
15   From the Select DDI Plan list select the DDI Plan required
16   In the Capacity field enter the number of connections to be allowed on this link.
17   Select Apply

General	SIP
Trunk Details (Virtual)	

Name	Virtual
Description	External Trunk
Presentation Number	123456
DDI Call Plan	Standard
Dial Plan	
Capacity	3
Reserve Incoming Capacity	0
Reserve Voice Capacity	0
Reserve Data Capacity	0
Port	5060
Out Of Service	

Update Apply Cancel

18   Select the SIP page
19   From the Compression Type list box select G729A8K

20   In the NAT Server field enter the IP address allocated by the ADSL provider

21   In the NAT Port field enter *5060*

22   From the NatMode field select Static

23   If a Registration Address, User Name and Password have been supplied by the SIP provider enter this information in the relevant fields otherwise leave the fields blank.
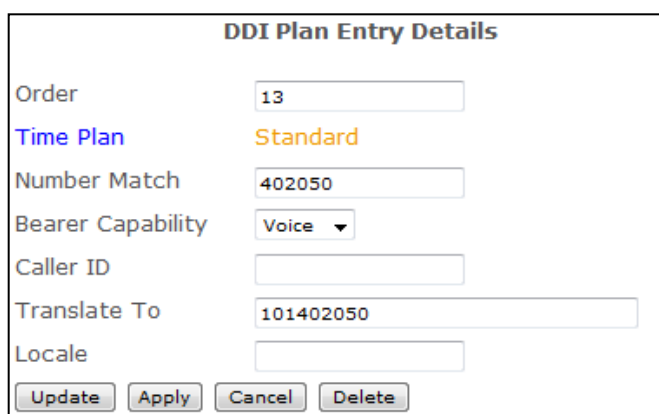
24   Select Update or Apply when ready.

Configuring a Trunk to Trunk call
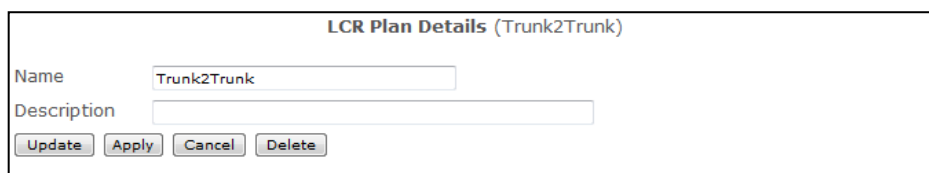
Incoming calls via a particular trunk(s) can be routed directly to another trunk where, for example, a video conference unit is connected. This facility can be configured as follows.

1   Open the DDI Call Plan assigned to the Trunk(s) receiving the incoming call.

2   Create a DDI Call Plan entry for the number being dialled to, for example, access the video conference unit.

3   In the Translate To field enter the DDI number prefixed with a number that does not exist on the system, similar to the following example.
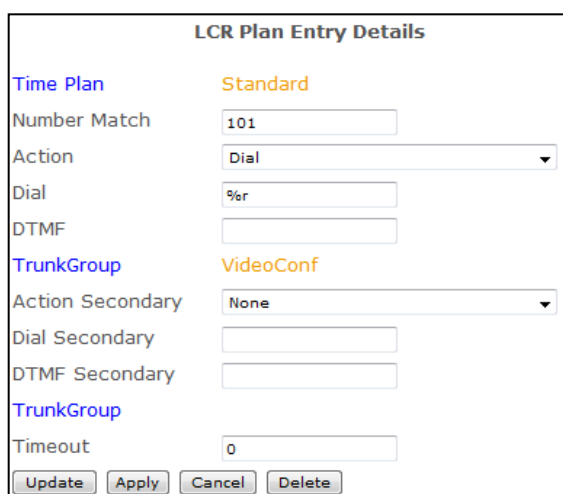
For further information on DDI Plans please refer to the Working with DDI Plans section from page 169.

4   When the system cannot find the number in the Translate To field it will look for a LCR Plan called "Trunk2Trunk". Therefore create a LCR Plan called Trunk2Trunk.

LCR Plan Details (Trunk2Trunk)

Name	Trunk2Trunk
Description	

Update   Apply   Cancel   Delete

5   Within this new LCR Plan create an entry that will route the call to the Trunk Group containing the trunk that the call is to be routed out on, for example, the Trunk that the video conference unit is connected to, similar to the following example.

LCR Plan Entry Details

Time Plan	Standard
Number Match	101
Action	Dial
Dial	%r
DTMF	
TrunkGroup	VideoConf
Action Secondary	None
Dial Secondary	
DTMF Secondary	
TrunkGroup	
Timeout	0

Update   Apply   Cancel   Delete

For further information on LCR Plans please refer to the Working with LCR Plans section from page 175.

Please note that if the Trunk2Trunk LCR Plan does not exist the caller will receive busy.

Working with DDI Plans

In order to direct your incoming DDI numbers to the correct User, Department, data connection etc a DDI Plan is used to route the calls to the correct location. By default, a Standard DDI Plan is created which routes all incoming external calls to the Main Department.

Creating a DDI Plan

1   In Manager select DDI Plans
2   Select the Add button
3   Enter a Name, using alpha-numeric characters only, do not use any punctuation and begin with a letter, and a Description if required
4   Select Apply

Alternatively, amend the default Standard DDI Plan..

Creating a DDI Plan entry

1   Select the Add DDI Entry button
2   If this entry is only to be active during certain hours select the Time Plan field and select the Time Plan required
3   In the Number Match field enter the DDI Number required. The number entered must be the number presented by the exchange (typically last 6 digits) as an exact match is made. This field is also relevant to outgoing CLI for the User as explained below
4   From the Bearer Capability list box select the option required eg Voice for connection to a User or Department
5   In the Translate To field enter the extension number or name of the User or Department to which calls to this DDI number are to be routed.
6   Select Update
7   Repeat steps 5 to 10 to add further entries
8   Select Update or Apply when ready.

It is recommended that a default entry be created in which the Number Match field is left blank, as shown by the first entry in the diagram above. This example will route all incoming calls to the Main Department unless a specific entry exists.

If a Time Plan is used the call will be routed during the hours specified by the Time Plan. Outside of those hours the system will look for the next match in the list or use the default entry.

Please note: if a DDI Plan entry is to be used to determine the outgoing CLI for a call the Translate To field must contain the User or Department's extension number only (please refer to page 171 for further details).

Please refer to page 243 for details on how to import or export DDI Plan entries.

Determining the order of a DDI Plan Entries
DDI Plan entries are read by the system from top to bottom of the list. The Order field can be used to determine the position of an entry in the list. The entries are then displayed in sequential order with low numbers at the top and higher numbers lower down. This is useful when, for example, calls to a specific number are to be routed to one location during a specific time of the day and then to another location outside of those hours. The DDI Plan entry containing the Time Plan should have a lower Order number than the alternative DDI Plan entry.

DDI Plan commands

The following prefixes can be entered in front of the entry in the Translate To field to provide the following functionality:

- (hyphen) = will place the dialled number into the caller's name field. This means that when a call is received the incoming CLI will be ignored and Caller Display/Call Status will display the DDI number the call was received on instead.

= (equal sign) = will record all calls received on this number. Please refer to the Call Recording section from page 232 for further information.

~ (tilde) = will bypass any compression (required for fax calls)

Please note that these options may cause the outbound CLI to fail therefore double entries will be required.

Outgoing CLI for external calls

A DDI Plan entry is also used when external calls are made by a User. If the extension number making the call is found in the Translate To field of a DDI Plan entry the number in the Number Match field will be used as the outgoing CLI. (Please note if the User's Dial Plan contains an entry specifying the outgoing CLI this will take precedence (please refer to page 118 for further details).) If no matches are found then the Trunk's Presentation Number will be used.
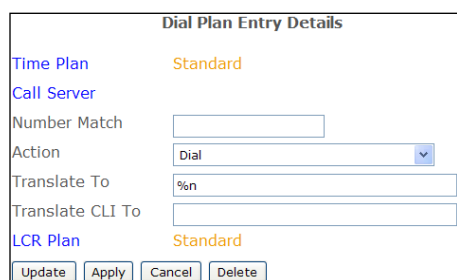
The order in which the configuration of the system will determine the outgoing CLI to be presented for a specific User when making an external call is as follows:
1   Dial Plan entry – Translate CLI To field
2   DDI Call Plan entry
3   Trunk – Presentation Number field

```
┌─────────────────────────────────┐
│ User dials from extension "x"   │
└─────────────────────────────────┘
                │
                ▼
┌─────────────────────────────────┐
│ Digits dialled are matched by a │
│ Dial Plan entry                 │
└─────────────────────────────────┘
                │
                ▼
┌─────────────────────────────────┐        No
│ Extension number "y" found in   │────────────────────┐
│ Translate CLI To field of the   │                    │
│ Dial Plan entry                 │                    │
└─────────────────────────────────┘                    │
                │ Yes                                   │
                ▼                                       ▼
┌─────────────────────┐   No   ┌──────────────────────────┐   No
│ Extension number    │───────▶│ Extension number "x"     │──────────────┐
│ "y" matched in DDI  │        │ matched in DDI Plan entry│              │
│ Plan entry          │        └──────────────────────────┘              │
└─────────────────────┘                   │ Yes                          │
          │ Yes                            ▼                              ▼
          ▼                    ┌──────────────────────────┐   ┌────────────────────┐
┌─────────────────────┐       │ DDI number in Number     │   │ Trunk Presentation │
│ DDI number in       │       │ Match field of DDI Plan  │   │ Number sent as CLI │
│ Number Match field  │       │ entry sent as CLI        │   └────────────────────┘
│ of DDI Plan entry   │       └──────────────────────────┘
│ sent as CLI         │
└─────────────────────┘
```

Examples

1

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	
Action	Dial
Translate To	%n
Translate CLI To	
LCR Plan	Standard

[Update] [Apply] [Cancel] [Delete]

A User on extension 2001 dials an external number, which is matched by the default Dial Plan entry. The Translate CLI To field is blank so no outgoing CLI has been specified. The LCR Plan determines the Trunk that this call will use. The DDI Plan assigned to this Trunk is checked for an entry for extension 2001. No entry exists therefore the Trunk's Presentation number is sent as the outgoing CLI.

2

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	
Action	Dial
Translate To	%n
Translate CLI To	
LCR Plan	Standard

[Update] [Apply] [Cancel] [Delete]

DDI Plan Entry Details

Order	1
Time Plan	Standard
Number Match	402001
Bearer Capability	Voice
Caller ID	
Translate To	2001
Locale	

[Update] [Apply] [Cancel] [Delete]

A User on extension 2001 dials an external number, which is matched by the default Dial Plan entry. The Translate CLI To field is blank so no outgoing CLI has been specified. The LCR Plan determines the Trunk that this call will use. The DDI Plan assigned to this Trunk contains an entry for the User, ie the Translate To field contains the User's extension number 2001. The outgoing CLI for this call will be 402001.

3

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	
Action	Dial
Translate To	%n
Translate CLI To	8000
LCR Plan	Standard

[Update] [Apply] [Cancel] [Delete]

DDI Plan Entry Details

Order	1
Time Plan	Standard
Number Match	402000
Bearer Capability	Voice
Caller ID	
Translate To	8000
Locale	

[Update] [Apply] [Cancel] [Delete]

A User on extension 2001 dials an external number, which is matched by the default Dial Plan entry. This Dial Plan entry has been configured to send out the DDI for the Main Department as the outgoing CLI. The LCR Plan determines the Trunk that this call will use. The DDI Plan assigned to this Trunk contains an entry for the Main Department, ie the Translate To field contains the Department's extension number. The outgoing CLI for this call will be 402000.

When configuring this facility it is recommended that Dial Plan entries similar to the following examples are created for all ranges of extension numbers used on the system to ensure that internal calls display the extension number calling rather the number entered in the Translate CLI To field in the above Dial Plan entry eg 8000.

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	2???
Action	Dial
Translate To	%n
Translate CLI To	
LCR Plan	Standard

[Update] [Apply] [Cancel] [Delete]

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	8???
Action	Dial
Translate To	%n
Translate CLI To	
LCR Plan	Standard

[Update] [Apply] [Cancel] [Delete]
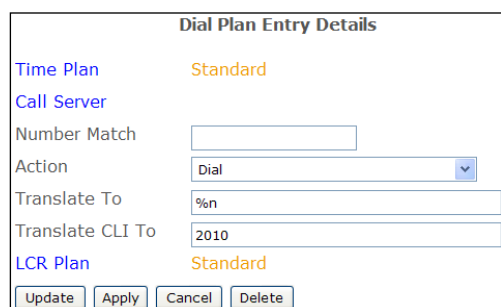
Please note: If Presentation Numbers outside the DDI range are to be used then a dummy entry of Number Match = <presentation number> and TranslateTo = <presentation number> must be added. This is to authorise the system to output a number not used within the DDI range.

Force outgoing CLI for Manager/Secretary working

The following Dial Plan entry gives an example of how to determine the outgoing CLI for, for example, a Manager who wish to give out his/her assistant's DDI number as CLI rather than their own.
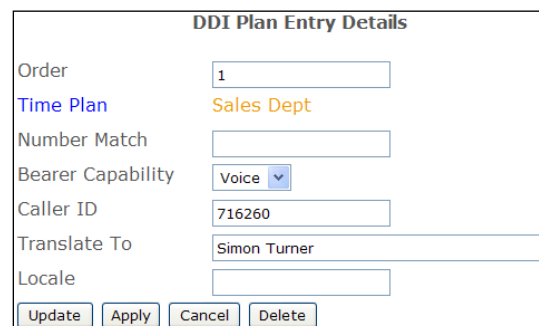
This Dial Plan entry would be created within a Dial Plan assigned only to the Manager. The Translate CLI To field contains the extension number of the Manager's assistant. This extension number must then be matched in the assistant's DDI Plan entry. The assistant's DDI number would then be used as outgoing CLI for all outgoing calls.

Routing calls dependent on the incoming CLI

Calls can be routed to a specific location when a certain incoming CLI is received by the system. For example, you may wish to route calls to a particular Department when calls are received from a specific customer. This feature can be configured as follows:

1   In Manager select DDI Plans
2   From the DDI Plan List select the DDI Plan required
3   Select Add DDI Entry
4   Enter the number required in the Order field.
5   If this entry is only to be active during certain hours enter the relevant Time Plan
6   In the Caller ID field enter the incoming number, eg the customer's telephone number. The digits dialled are matched from the left and the system will look for closest match.
7   In the Translate To field enter the extension number or name of the User or Department to which calls are to be routed.
8   Select Update or Apply when ready.
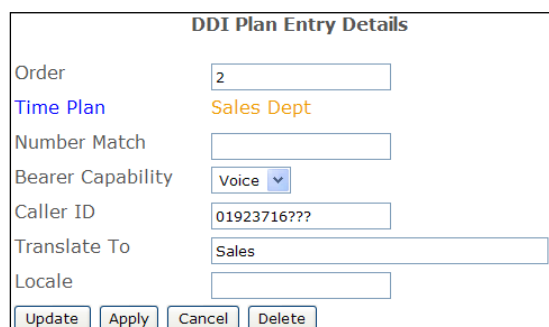
Routing calls from a DDI range

The Caller ID field has the ability to left hand partial match numbers, using ?'s, which means, for example, that a DDI range (eg 01923716200-299) could be matched and the call presented to a User or Department, as per this example.

Routing calls when the incoming CLI has been withheld

DDI Plan Entry Details

Order	1
Time Plan	Standard
Number Match	
Bearer Capability	Voice
Caller ID	?
Translate To	Reception
Locale	

Update Apply Cancel Delete

A question mark (?) entered in the Caller ID field, as shown by this example, will route calls where the incoming CLI has been withheld.

Routing calls from a Contact to a specific User or Department

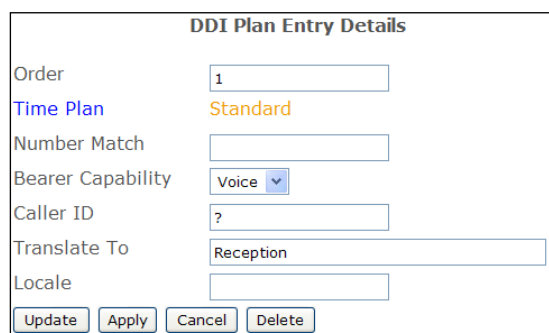This feature uses the Direct Route To field within a Contact, for further information on Contacts please refer to page 141. However the system needs to know which Company's list of Contacts to search in order to use this feature. Therefore please refer to page 158 within the Creating Companies section for full details.

Working with LCR Plans

Least Cost Calls Routing Plans (LCR Plans) determine the routing of outgoing external calls; this may be simply to specify the Trunk Group to be used or to specify the carrier that will take the call. By default, a Standard LCR Plan is created which will route all external calls via the AllTrunks Trunk Group.

It is recommended that an LCR plan is also used for facilities going to or not going to a Trunk such as call barring, dialling a prefix and so on. Example Dial Plan entries are available within the Working with Dial Plans section from page 114.

Creating a LCR Plan
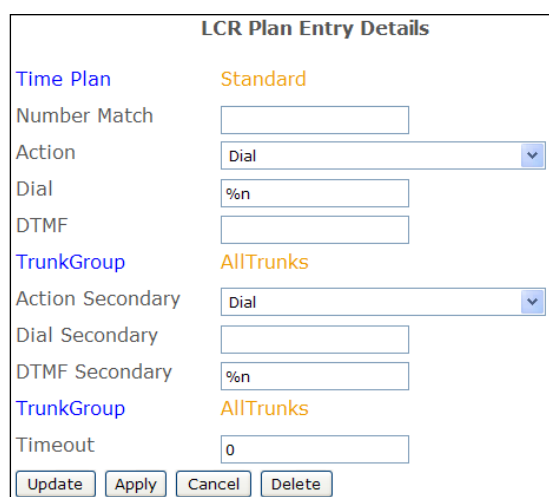
1   In Manager select LCR Plans
2   Select Add
3   Enter a Name, using alpha-numeric characters only, begin with a letter and do not use any punctuation, and a Description if required
4   Select Apply
5   Select the Add LCR Plan Entry button
6   If this entry is only to be active during certain hours enter the relevant Time Plan (see page 148 for further information).
7   Select Apply if a change has been made to the Time Plan field
8   In the Number Match field enter the number to be dialled by eg the User
9   From the Action list box select the Action required (please refer to page 294 for a full list of the functions of each Action)
10   In the Dial field enter the number to be dialled by the system, and/or
       a   If all the digits entered, by eg the User, are to be used enter *%n*
       b   If the digits entered minus the numbers in the Number Match field are to be used enter *%r*
       c   To use the User's extension number enter %e

11   In the DTMF field enter the digits to be dialled once a connection has been made. This may be a pass code required by the carrier. A pause may be required either before or after the pass code in which case insert a D either before and/or after the code digits.

12   Select Apply

13   Select the Trunk Group field and select the Trunk Group that calls via this LCR Plan entry will use (see page 159 for further details).

14   If required repeat steps 9 to 13 using the Action Secondary field, Dial Secondary field, DTMF Secondary field, secondary Trunk Group field. The Secondary entries will be used if the primary entries fail.

15   Select Update when ready.

16   Repeat steps 5-15 to enter further LCR Plan entries if required.

17   Select Update or Apply when ready.

18   This new LCR Plan can now be assigned to a Dial Plan entry

Alternatively, amend the default Standard LCR Plan and follow steps 5 to 17.

An understanding of the Working with Dial Plans section from page 114 will help with the completion of LCR Plan entries. It is recommended that Dial Plans are used for internal dialling and LCR Plans are used for external calls (including call barring).

Examples

LCR Plan Entry Details

Time Plan	Standard
Number Match	
Action	Dial
Dial	%n
DTMF	
TrunkGroup	AllTrunks
Action Secondary	Dial
Dial Secondary	
DTMF Secondary	%n
TrunkGroup	AllTrunks
Timeout	0
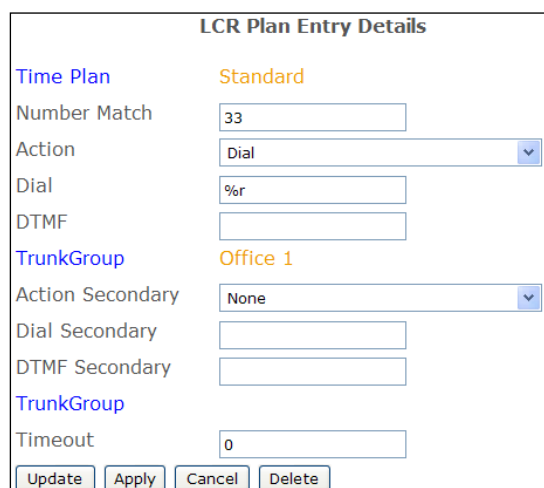
Update Apply Cancel Delete

This diagram gives an example of a default LCR Plan entry which will route external calls to any available Trunk.

LCR Plan Entry Details

Time Plan	Standard
Number Match	33
Action	Dial
Dial	%r
DTMF	
TrunkGroup	Office 1
Action Secondary	None
Dial Secondary	
DTMF Secondary	
TrunkGroup	
Timeout	0

Update Apply Cancel Delete

This diagram gives an example of a LCR Plan entry which will route calls to a specific Trunk(s) via a prefix.

This diagram gives an example of a LCR Plan entry, which is used to route calls via a specific carrier.

Using Area Codes

Where an incoming CLI match is not found within a Contact it may be useful to display where the call has originated from. Area Codes can be created to match the area code part of an incoming number to display the town/city that the caller is ringing from, for example 01923123456 could be displayed as Watford 123456. Please note that incoming CLI matched by a Contact will have priority over an Area Code entry. An Area Code can be created as follows:

1   In Manager select Area Codes
2   Select the Add button
3   Enter the Name of the area to be displayed, using alpha-numeric characters only, begin with a letter and do not use any punctuation, eg Watford
4   In the Ambiguous Telephone Number field enter the number to be matched against incoming CLI, using a question mark (?) where a digit will be variable, eg 01923??????.
5   Select Update or Apply when ready.

Please note: The number of digits must match the number of possible digits; some exchanges may require multiple entries.

A list of ready-made UK Area Codes is available with the **max**imiser system and can be imported into the configuration. Please refer to page 240 for further details.

Configuring Music Channels

The Music Channels section of the configuration allows you to determine the music or message to be played to an external caller while their call is on hold. A Call Server can support up to 6 Music Channel entries allowing a different message or piece of music can be played to callers to a specific Company, Department or User.

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Music can be stored on the Call Server as a WAV file or supplied via an external music source. The ability of a specific Call Server to support this facility is controlled by the relevant Voicemail Port. Within this Voicemail Port ensure that Music Server option is ticked. Please refer to the Voicemail Ports section from page 188 for further details.

Please note: A music channel will not use a voicemail channel.
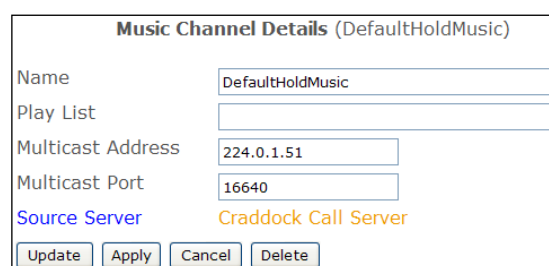
Internal Music on Hold

The music/message to be played must be stored as a WAV file (CCITT A-Law 8.000kHz 8bit mono) on the relevant Call Server. (Microsoft's Sound Recorder will allow you to format a "normal" WAV file to this format or Goldwave is available at a modest cost (www.goldwave.com).) There is no limit to the number and size of WAV files that can be stored on the Call Server (other than disk space). (Please note that 8000Hz mono 16bit (128kbps) WAV files can also be used but they are twice the size.)

Please note: WAV files are a general resource. If used for messages callers will not be automatically played the message from the beginning.

A default Music Channel entry called DefaultHoldMusic will play the default WAV file provided with the system.

The Play List field is blank, which indicates that the default WAV file should be used. This entry will play "Please hold" to all external callers placed on hold; however this default configuration can be changed as follows:

Music Channel Details (DefaultHoldMusic)

Name	DefaultHoldMusic
Play List	
Multicast Address	224.0.1.51
Multicast Port	16640
Source Server	Craddock Call Server

[Update] [Apply] [Cancel] [Delete]

Change the Default Music on Hold

The default Music Channel can play an alternative wav file if required. This file must be stored in the correct format and stored in the ftp directory of the Call Server as follows.
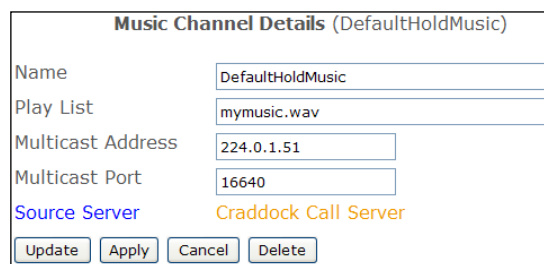
1   Format the WAV file to be used as CCITT A-Law 8.000kHz 8bit mono.
2   Using FTP via a browser or Command Prompt log in as Admin on the relevant Call Server as described in the Admin Access to a Call Server section from page 252.

If you are using a browser, copy the wav file into this area.

If you are using the Command Prompt:
a   Place the WAV file in the root, eg C:\
b   In the Command Prompt enter *put <filename>.wav*, eg put mymusic.wav, to copy the file from root to the ftp directory.
c   After receiving the Transmission Complete message enter *Quit*

3   In Manager, select Music Channels
4   Open the DefaultHoldMusic entry by selecting
     the ID field
5   In the Play List field enter the name of the WAV
     file, eg mymusic.wav

Music Channel Details (DefaultHoldMusic)

Name	DefaultHoldMusic
Play List	mymusic.wav
Multicast Address	224.0.1.51
Multicast Port	16640
Source Server	Craddock Call Server

[Update] [Apply] [Cancel] [Delete]

The Source Server field indicates the Call Server where the WAV file is stored and that calls to this Call Server will be played this music/message. This is useful where multiple Call Servers are in operation on one system and a different music/message should be played at each Call Server. If this field is left blank the WAV file will be played for calls received on all Call Servers operational on the system however the WAV file must be stored on each Call Server.

Create a Music Channel for a Department, User or Company

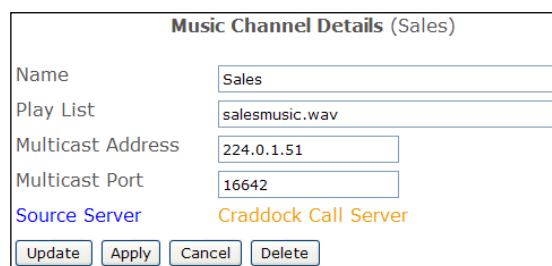To use a different WAV file for a specific Department, User or Company, create a Music Channel entry similar to the following example:

1   Format the WAV file to be used as CCITT A-Law 8.000kHz 8bit mono.
2   Use FTP to send this WAV file to the ftp directory on the Call Server as detailed above
3   In Manager, select Music Channels
4   Select the Add button
5   In the Name field enter the name of the
     Department, User or Company
6   In the Play List field enter the name of the WAV
     file copied to the Call Server in Step 2
7   In the Multicast Address field enter 224.0.1.51
8   In the Multicast Port field enter an even number
     that you have not used in another Music
     Channel entry, eg 16642, (not less than 1024).
9   Select Apply
10   Select the Source Server field
11   From the Select Call Server list select the Call Server where the WAV file is stored. The
     music/message will be played to the calls received by this Call Server. If left blank the WAV file will
     be played for calls received on all Call Servers operational on the system however the WAV file
     must be stored on each Call Server.
12   Select Update or Apply when ready.

Music Channel Details (Sales)

Name	Sales
Play List	salesmusic.wav
Multicast Address	224.0.1.51
Multicast Port	16642
Source Server	Craddock Call Server

[Update] [Apply] [Cancel] [Delete]
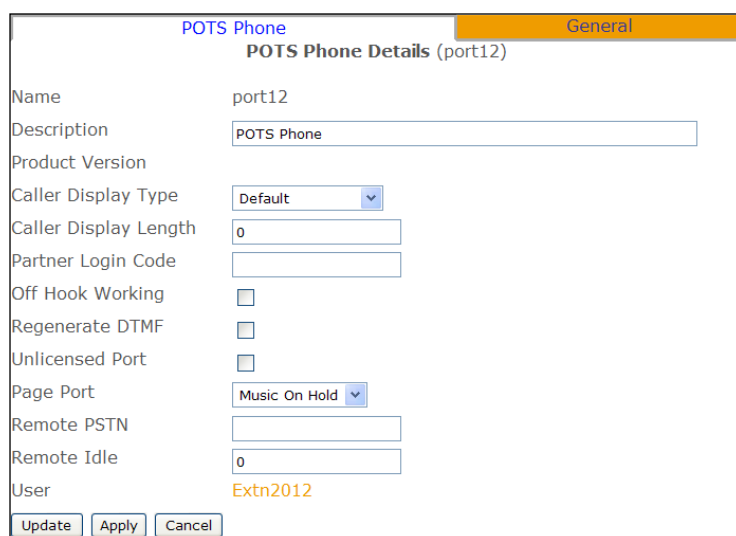
If a Music Channel entry has been created for a Company this WAV file will be used for all external calls into this company unless there is an entry specific to a User or Department within that Company.

External Music on Hold

Music on hold can be supplied into the system via an analogue phone port. You will require a "Music on Hold" Adapter, which converts a 3.5 mm stereo jack to RJ45 for connection into a phone module port. (Technical details of the adapter can be found on page 339.) The music source should provide approx 1V p-p and have its own volume control; the volume cannot be adjusted via the Call Server or Phone Module. Only one music source can be connected to an analogue port. If multiple music sources are required plug each one into a separate analogue phone port.

To use an external source the following configuration in Manager is required. Firstly the analogue phone port must be configured to accept an external music source.

1. In Manager, select Phones
2. Select the Call Server or Phone Module required
3. From the Phone List select the port required
4. From the Page Port list box select Music on Hold
5. Select Update or Apply when ready.

POTS Phone	General

POTS Phone Details (port12)

Name	port12
Description	POTS Phone
Product Version	
Caller Display Type	Default
Caller Display Length	0
Partner Login Code	
Off Hook Working	☐
Regenerate DTMF	☐
Unlicensed Port	☐
Page Port	Music On Hold
Remote PSTN	
Remote Idle	0
User	Extn2012

[Update] [Apply] [Cancel]

Finally the relevant Music Channel must be configured to use the analogue phone port as the source for the hold music. To use the external source as the default music on hold source amend the DefaultHoldMusic entry as follows:

1. In Manager, select Music Channels
2. Open the DefaultHoldMusic entry by selecting the ID field.
3. In the Play List field enter any text, eg CD player on port 12. This must not be the same as the name of a WAV file currently stored on the Call Server.
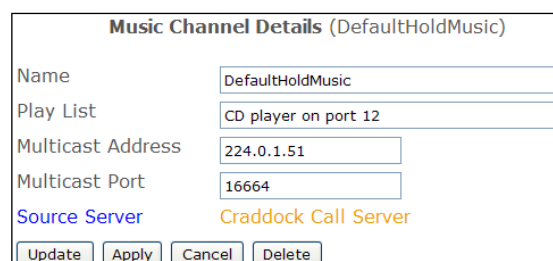4. Edit the Multicast Port field to equal 16640 + (2 x port number), eg for port 12 the entry will be 16664.
   (If more than one Phone Module exists on the system multicast will find the relevant port.)
5. Where more than one Call Server is operational within a system leave the Source Module field blank to play the music globally alternatively enter a specific Call Server to play the music only for calls received on that Call Server.
6. Select Update or Apply when ready.
7. Reboot the relevant Call Server/Phone Module if required.

Music Channel Details (DefaultHoldMusic)

Name	DefaultHoldMusic
Play List	CD player on port 12
Multicast Address	224.0.1.51
Multicast Port	16664
Source Server	Craddock Call Server

[Update] [Apply] [Cancel] [Delete]

If you wish the external music to be played for calls to a specific Department, User or Company create a new Music Channel with the same entries as described above and enter the Department, User or Company name in the Name field.

To ensure that calls are not made to the "music on hold" port it is recommended that the User assigned to this port has Do Not Disturb set and voicemail is not enabled.
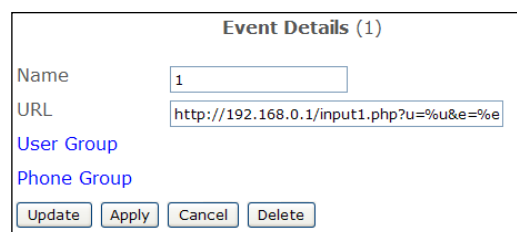
Please note: it is your responsibility to make your own arrangements with the Performance Rights Society (www.prs.co.uk) regarding licences for playing regular CD tracks.

Using Events

Events allow web pages to be sent to a PCS 60, 50 or 410/400 to display a message, for example, alert Users of a fire alarm, inform Users of maintenance on the network, etc.

Closing an external input contact or dialling a Dial Plan entry can activate an Event. An Event will only display the web page on a PCS 60, 50 or 410/400 connected to the Call Server where the Event was generated.

By default, the configuration contains 100 pre-configured Events. Event 0 will display each User's Home Page, Event 1 will display input1.php, and similarly Event 2 to 99 will display input2.php, input3.php etc. These files are stored on the internal Web Server and can be amended as required. The example pages will switch back to the User's previous page after 30 seconds; this is a result of the programming of these pages and not of the system.

Event Details (1)	
Name	1
URL	http://192.168.0.1/input1.php?u=%u&e=%e
User Group	
Phone Group	
Update Apply Cancel Delete	
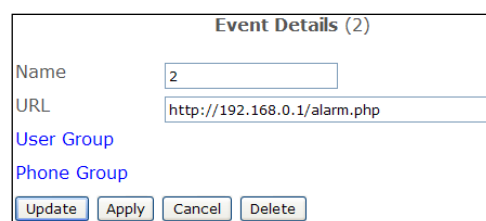
Please refer to the Internal Web Server section from page 184 for details on how to accessing these web pages.

Amending an existing Event

An existing Event can be amended to display the required web page when the Event is activated.

1 In Manager select Events
2 From the Events List select the event to be amended
3 In the URL field enter the address to the web page to be displayed when this event is activated. If this field is left blank the Event will display the User's Home Page.
4 Select Update or Apply when ready.

Event Details (2)	
Name	2
URL	http://192.168.0.1/alarm.php
User Group	
Phone Group	
Update Apply Cancel Delete	

Activating an Event

An Event can be activated by either:

1 A Dial Plan entry, as per the following example.

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	33
Action	Set Event
Translate To	%r
Translate CLI To	
LCR Plan	Standard

[Update] [Apply] [Cancel] [Delete]

This example Dial Plan entry can be used to activate all Events. Dialling 131 would activate Event 1, 132 would activate Event 2 and so on.

2   Relay Input ports. Closing a contact on the 2 input ports on the rear of the Call Server will activate Events 1 or 2.   Please refer to page 331 for further information.

Specifying the User Group

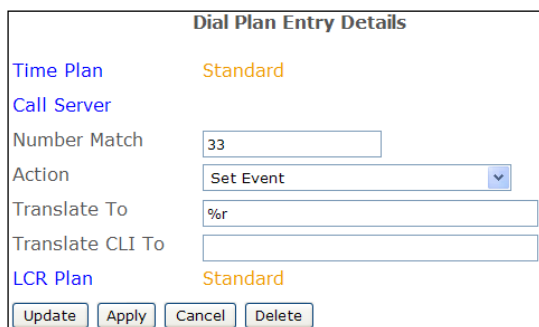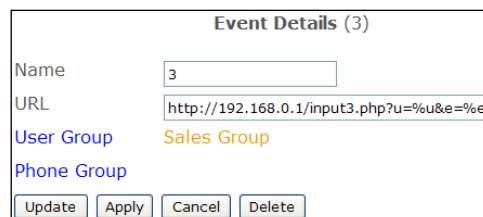Rather than sending an Event to all users of a PCS 60, 50 or 410/400 it may be necessary to send the web page to a specific User or group of Users. A Group containing this User or Users can be created and then the Event configured to send the web page only to the members of this Group. The User Group field within an Event is configured to use this feature as follows:

1   Create a Group containing the Users to be sent the Event. (Please refer to Creating a group of Users on page 122 for further details.)

2   In Manager select Event

3   From the Events List select the event to be amended

4   Select the User Group field

5   From the Select Group list select the Group created above

6   Select Update or Apply when ready.

7   This Event will now only be sent to the User's in the specified Group

Event Details (3)

Name	3
URL	http://192.168.0.1/input3.php?u=%u&e=%e
User Group	Sales Group
Phone Group	
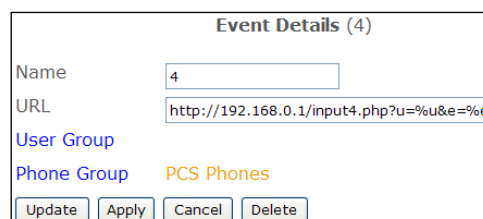
[Update] [Apply] [Cancel] [Delete]

Specifying the Phone Group

The Phone Group field within an Event can be used to specify the particular PCS 60, 50 or PCS 410/400s that should be sent the Event. This means that regardless of which User is currently logged on to a particular phone the message will be received. This feature can be configured as follows:

1   Create a Group containing the PCS 60, 50 or 410/400s to be sent the Event. (Please refer to the Creating a Group of Phones section on page 124 for further details.)

2   In Manager select Event

3   From the Events List select the event to be amended

4   Select the Phone Group field

5   From the Select Group list select the Group created above

6   Select Update or Apply when ready.

7   This Event will now only be sent to the phones in the specified Group

Event Details (4)

Name	4
URL	http://192.168.0.1/input4.php?u=%u&e=%e
User Group	
Phone Group	PCS Phones

[Update] [Apply] [Cancel] [Delete]

Using the TAPI Driver

The TAPI Driver available with the **max**imiser system enables users to dial directly from their Microsoft Outlook Contacts. This driver is currently available for Microsoft Windows 2000 and XP.

1   Download the TAPI driver from the SpliceCom Technical Forum
2   Unzip the file and save in \WINNT\System32 on the user's PC
3   On the user's PC, from Control Panel open Phones and Modem Options
4   Select the Advanced tab and select Add
5   Select **max**imiser TAPI Service Provider and select Add
6   With **max**imiser TAPI Service Provider selected, select Configure
7   In the IP Address field enter the IP Address of the Call Server or Phone Module
8   In the Port field enter the Port number to which the user's handset is connected, eg 5001, 5020 etc (this will be truncated to 1, 20 etc)
9   In the Access Code field enter the port's Partner Login Code
10   Select OK and then Close
11   Within Microsoft Outlook open the user's Contacts
12   Select the Dial icon
13   Select Dialling Options
14   From the Connect Using Line list box select **max**imiser on
15   Select OK and Close
16   The user will now be able to dial directly from a Microsoft Outlook Contact.

Please note:
- The IP address of the Call Server or Phone Module can be viewed within Manager. Select Modules and then select the relevant Call Server or Phone Module, the IP address is displayed.
- Each port on a Call Server or Phone module is given a reference number starting at 5001. Therefore port 1 is referenced as 5001, port 2 as 5002 etc.
- The port's Partner Login Code can be configured within Manager. Open the configuration form for the relevant User. Select the entry within the Initial Phone field and amend the Partner Login Code field as required.

Internal Web Server

A Web Server (Apache with PHP and LDAP module) is contained on the hard disk within the Call Server. This Web Server controls the PHP images displayed on the PCS 60, 50 and 410/400.

Accessing the Web directory

PHP pages can be transferred to and from the Web directory on the Call Server as follows:

Via the Command prompt on a PC connected to the system
1. Enter ftp <ip address>, eg ftp 192.168.0.1
2. At the User prompt enter *web*
3. At the Password prompt enter the Web Password, eg password

   Use the get command to copy a file from the Web directory to root, eg C:/
   eg get <filename>.php

   Use the put command to copy a file from root, eg C:/ to the Web directory
   eg put <filename>.php

4. The Transmission Complete message will be received when the action is complete
5. Enter Quit to exit.

or

Via your browser,
1. Enter ftp://web:<web password>@<ip address>, eg ftp://web:password@192.168.0.1
2. Copy and paste the required files to and from this area

For information on how to change the password used to access to the Web directory please refer to the System Passwords section from page 251.

Home Page

Each User can be configured with a Home Page, this will be the default screen displayed on their PCS 580, 60, 50 and/or 410/400. This can be an external web page if an Internet connection is available, or a page stored on the customer's Intranet, or a page stored on the internal Web Server. To point a User's Home Page to a file stored on the internal Web Server configure the following:

1. In Manager select Users
2. Select the User required
3. Select the Details page
4. In the User's Home Page field enter

   http://<ip address of Call Server>/<file name>.php.

5. Select Update or Apply when ready.

By default each User's Home Page field points to http://192.168.0.1/index.php. Index.php can be accessed as explained above and amended as required.

Using an Auto URL

A PCS 60, 50 and 410/400 can display web pages when the incoming CLI of a call matches a number entered in a Contact or when a call is routed via a Department. The path to the web page should be entered in the Auto URL field, this can point to an external web page if an Internet connection is available, or to a page stored on the customer's Intranet or to a page stored on the internal Web Server.

To point the Auto URL field to a file stored on the internal Web Server enter:

> http://<ip address of Call Server>/<file name>.php

Auto URL	http://192.168.0.1/contact.php
Auto URL Mode	On Connect
Auto URL Direction	Both

> eg http://192.168.0.1/contact.php

The Auto URL Mode field then determines when this web page will appear, either:

> On Ring – when a call is presented to the phone and before it has been answered.
> or
> On Connect – once the call has been answered.

Contact.php is an example page provided by default on the Web Server to demonstrate information available from the Call Server. This information could be used by a web author to, for example, search a database and display the results of the query.

To demonstrate the information available, in the Auto URL field, enter

http://<ip address of call server>/contact.php?n=%n&c=%c&u=%u&e=%e&d=%d&p=%p&t=%t

%n = Number	Caller's number
%c = Contact	Caller's name (if available)
%u = User Name	Recipient's User name
%e = Extension Number	Recipient's extension number
%d = Destination Number	Destination of the call – User or Department number
%p = Port	Address of the phone receiving the call
%t = Target Name	Name of the destination – User or Department name.

Events

An Event can display an external web page if an Internet connection is available, or a page on the customer's Intranet or a page on the internal Web Server.

To point an Event to a file stored on the internal Web Server enter:

> http://<ip address of Call Server>/<file name>.php

> eg http://192.168.0.1/input6.php

Please refer to page 181 for further information on Events.

Custom Script

The Custom Script option available in Manager via Utilities can be used to open a customised web page designed to display information to help you administer the **max**imiser system, for example, to display a list of Users with DND set.

1. Rename the web page created to custom.php
2. Use FTP to copy custom.php into the Web directory as described in the Accessing the Web Directory section above.
3. Telnet on to the Call Server as described in the Root Access to the Call Server section from page 270.
4. Enter

   cp /Web/custom.php /WebManager/

   This command will move the file to the correct location so that when the Custom Script option in Utilities is selected custom.php will be displayed.

Back up the web server

A back up of the web pages stored in the Web directory will be required to ensure that when, for example, a faulty Call Server is replaced all the web pages used by the system are restored on the new Call Server. This can be done by running the following back up script.

1. Telnet on to the Call Server via Root as described in the Root Access to a Call Server section on page 252.
2. Type /ftp/backup
3. This will create a file called web.tar within the ftp directory.
4. Using FTP via a browser or Command Prompt log in as Admin on the relevant Call Server as described in the Admin Access to a Call Server section from page 252.
5. If you are using a browser the web.tar file can be copied or moved to any location as required.
   or
   If you are using the Command Prompt use the "get" command to download the required file, eg get web.tar, and place it in the root, eg C:\. The Transmission Complete message will be received when the transfer is complete. Enter Quit when finished.

This back up script will also back up the internal voicemail server, please refer to the Maintaining and Troubleshooting Voicemail section from page 237 for further details.

Restore the web server

1. Copy the web.tar file into the Web directory on the Call Server as described in the Accessing the Web directory section above.
2. Telnet on to the Call Server via Root as described in the Root Access to a Call Server section on page 252.
3. Change to the Web directory, ie cd /Web
4. Type tar –xvf web.tar
5. This will expand all the web pages into the Web directory.

Working with Voicemail

Embedded voicemail

Embedded voicemail is run from a 5100 Call Server, 5108 Call Server, 4100 Call Server or 4140 Remote Call Server and provides standard voicemail and auto attendant facilities. All configuration is performed via Manager.

- A 5100 Call Server provides up to 16 simultaneous connections, 4 of which are licensed on shipment, and 1500 hours of message storage
- A 5108 Call Server provides up to 8 simultaneous connections, 2 of which are licensed on shipment, and 500 hours of message storage
- A 4100 Call Server provides up to 8 simultaneous connections and 30 hours of message storage
- A 4140 Remote Call Server provides up to 4 simultaneous connections and 10 hours of message storage

Voicemail messages are stored on the internal hard disk and require 0.5 MB of disk space for each 1 minute of message.

Voice Processing Application

The voicemail and auto-attendant application can also be run on a standalone Linux PC or server.   This is primarily for scenarios where more than 16 concurrent channels are required and/or greater storage capacity is required. A Voice Processing Port licence is required to support this facility, one licence per port required. Therefore, for example, if 24 concurrent Voice Processing ports are required, 24 x Voice Processing Port licences must be purchased. Please refer to the Installing Licences section from page 35 for further information.

When running on a networked Linux PC or Server, the **max**imiser Voice Processing application has been tested running up to 120 concurrent voicemail ports when hosted on an HP 2.8 GHz Pentium 4 PC running 512 MB of memory.

Please note that the maximiser Voice Processing application is responsible for running the following;
- Voicemail
- Auto Attendant
- IVR (ESP)
- Call Recording
- Meet Me Conferencing
- Queue Entry & In-Queue Announcements
- Music-on-Hold

As such, all these tasks will dynamically consume Voice Processing channels. Please take this into account when planning capacity for your requirements.
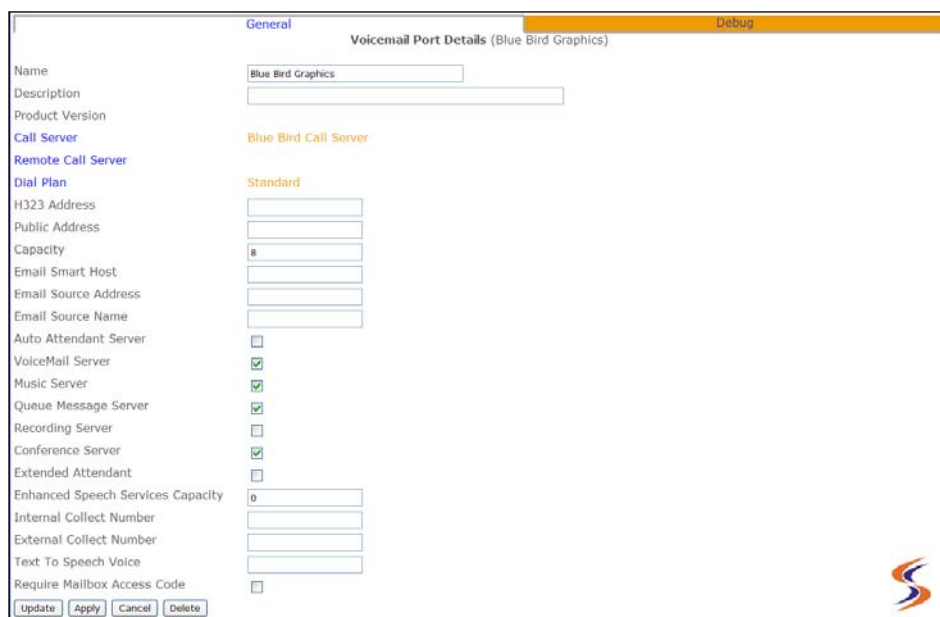
Please refer to the "Linux for **max**imiser – An Introduction" document for installation instructions. This is available within the Partner area of the SpliceCom website.

Voicemail Ports

A Voicemail Port determines the facilities to be provided by a Call Server's voicemail service. By default, one Voicemail Port is created on the database and can be used to control the entire voicemail facility. However on systems with more than one Call Server additional Voicemail Ports can be created if different settings are required for each Call Server, for example, when Call Servers need to connect to different email servers for the Voicemail Email facility.

A Voicemail Port can be created as follows:

1   In Manager select Voicemail Ports
2   Select the Add button
3   Enter a Name for the Voicemail Port, using alpha-numeric characters only, begin with a letter and do not use any punctuation. If this Voicemail Port will be using the internal voicemail on the relevant Call Server the Name must be "BuiltIn" (case sensitive) otherwise any alpha-numeric characters can be entered.
4   Enter a Description if required.
5   Select Apply
6   Select the Call Server field.
7   From the Select Call Server list select the Call Server providing the voicemail service. If left blank, this Voicemail Port will be used by the entire system.
8   Select Apply
9   Select the Dial Plan field.
10 From the Select Dial Plan list select the required Dial Plan. This will determine how digits dialled by the voicemail server are handled. This is required when using Auto Attendant, Call Back, Ring back when free and Assistant Telephone functionality.
11 In the Capacity field enter the number of Voice Processing Port licences to be assigned to this port. This will be the maximum number simultaneous connections that will be allowed via this port.
12 Tick Voicemail Server so that this Call Server will support voicemail functionality.
13 If this Call Server is to support Auto Attendant functionality tick Auto Attendant Server. (Please refer to the Creating an Auto Attendant section from page 222 for further information.)
14 If this Call Server is to support Music on Hold tick Music Server. (Please refer to the Configuring Music Channels from page 177 for further information.)
15 If this Call Server is to support Department Announcements tick Queue Message Server. (Please refer to the Voicemail for a Department section from page 209 for further information.)
16 If this Call Server is to support call recording tick Recording Server. (Please refer to the Call Recording section from page 232 for further information.)
17 If this Call Server is to support the conferencing feature tick Conference Server. (Please refer to the Conferencing section from page 151 for further information.)
18 Select Update or Apply when ready.

Licensing

MessageBox licence

Each User and Department on the system can be configured to use the voicemail functionality. To activate this use a MessageBox licence must be purchased and assigned to the User or Department.

When purchasing a Call Server the following number of MessageBox licences are provided.
- A 5100 Call Server will be provided with 10 MessageBox licences
- A 5108 Call Server will be provided with 5 MessageBox licences
- A 4100 Call Server will be provided with 30 MessageBox licences,
- A 4140 Remote Call Server is provided with 10 MessageBox licences.

Additional MessageBox licences can be purchased up to a maximum of 1,000 per system or 50 on a standalone 5108 Call Server or 100 on a standalone 4140 Remote Call Server.

A list of which Users and Departments have been assigned MessageBox licences can be viewed in Manager by selecting Utilities and then Mailbox Licences.

Voice Processing Port Licence

Each 5100 Call Server is supplied with 4 simultaneous connections to voicemail however the 5100 can support up to 16 connections. Each 5108 Call Server is supplied with 2 simultaneous connections to voicemail however the 5108 can support up to 8 connections. Therefore to enable the additional connections a Voice Processing Port licence must be purchase. One licence per port/connection.
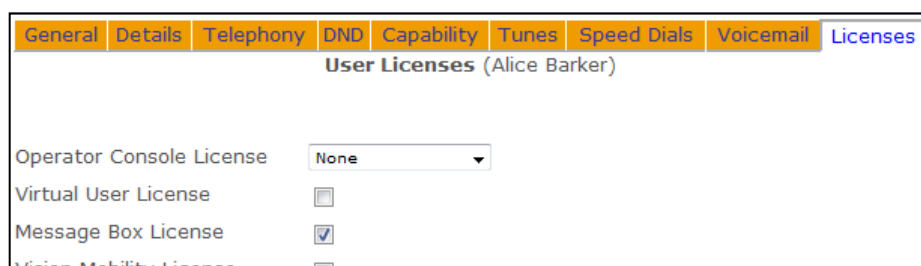
This licence will also facilitate the use of the voicemail application run on a standalone Linux PC or server.

For further information please on licensing refer to the Installing Licences section from page 35.

Enabling Voicemail for a User

In order to provide a User with voicemail functionality a Message Box Licence must be assigned to that User.

   1   In Manager select Users
   2   From the Users list select the User required
   3   Select the Licences page
   4   Tick the Message Box License field
   5   Select Update or Apply when ready.

If there are insufficient licences available this setting will not be saved. A list of which Users have been assigned MessageBox licences can be viewed in Manager by selecting Utilities and then Mailbox Licences.

Once a MessageBox licence has been assigned the User will have use of the following facilities:
• Listen to messages (as described in the Accessing Voicemail section from page 193).
• Colleagues will be able to transfer calls to the User's voicemail (as described in the Leaving a Message section from page 192).
• Call recording (as described in the Call Recording section from page 232).

Further voicemail functionality can be provided to a User by configuring their Voicemail Enabled option as follows:

   1   In Manager select Users
   2   From the Users list select the User required
   3   Select the Voicemail page
   4   Tick the Voicemail Enabled field
   5   Select Update or Apply

General	Details	Telephony	DND	Capability	Tunes	Speed Dials	Voicemail	Licenses

User Voicemail (Alice Barker)

Voicemail Enabled	☑

Once this option has been enabled callers to a User's extension will be automatically transferred to voicemail if the User's extension is busy or not answered as described in the Leaving a Message section from page 192.

Specifying the Call Server providing voicemail functionality

The Current Home field within the User's General page determines which Call Server is providing the User with the voicemail service.   This is relevant when a system consists of multiple Call Servers and it is necessary to know which Call Server is storing the User's messages and to ensure the correct Call Server is being used.

General	Details	Telephony	DND	Capability	Tunes	Speed Dials	Voicemail	Licenses

User General (Alice Barker)

Name	Alice Barker
Description	Users Description
Telephone Number	2006
Initial Phone	Craddock Call Server.port06
Locale	
Company	
Current Home	Craddock Call Server

Configuring the Voicemail Enabled field via a Dial Plan entry

A User's Voicemail Enabled option can be enabled or disabled via Manager as described above or via a Dial Plan entry.

The following diagrams give example Dial Plan entries that can be created to enable or disable a User's Voicemail Enabled field.

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	*40
Action	Voicemail Enable
Translate To	
Translate CLI To	
LCR Plan	Standard

Update   Apply   Cancel   Delete

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	*41
Action	Voicemail Disable
Translate To	
Translate CLI To	
LCR Plan	Standard

Update   Apply   Cancel   Delete

The Translate To fields are empty as the Dial Plan entry will only enable or disable the Voicemail Enabled option for the User on whose handset this short code is dialled. On dialling these short codes a recorded message will be played confirming that this option has been enabled or disabled.

For further information please refer to the Working with Dial Plans section from page 114.

Leaving a Message

If a User's Voicemail Enabled field has been ticked callers to a User's extension or DDI number will automatically be transferred to voicemail if the User's extension is busy or not answered within the User's No Answer Time.

Callers will be played the default message – "No one is available to take your call at the moment. Please leave a message after the tone." followed by a beep. Callers will be given 5 minutes to record a message. If no sound is heard for 10 seconds the caller will be automatically disconnected.

A caller can interrupt the greeting by pressing #.

If the caller hangs up without leaving a message the User will not receive a blank message.

Transfer to Voicemail

If a User wishes to be able to dial directly to another User's voicemail or to transfer a caller to another User's voicemail this can be done either via their PCS 580/570/560, PCS 410/400, PCS 100, PCS 60 or PCS 50, or via a Dial Plan entry similar to the following examples:

Dial Plan Entry Details		or	Dial Plan Entry Details	
Time Plan	Standard		Time Plan	Standard
Call Server			Call Server	
Number Match	#2001		Number Match	#
Action	Dial		Action	Dial
Translate To	!2001		Translate To	!%r
Translate CLI To			Translate CLI To	
LCR Plan	Standard		LCR Plan	Standard
Update Apply Cancel Delete			Update Apply Cancel Delete	

The exclamation mark (!) in the Translate To field = Go to Voicemail.

Using the Assistant Telephone Number

Each User can be configured with an internal or external number to which callers can be transferred if they do not wish to leave a message. Pressing 0 during or after the User's greeting activates the Assistant Telephone number. The User will need to amend their greeting to inform callers that this facility is available.

This facility can be configured as follows:

1   In Manager select Users
2   From the Users list select the User required
3   Select the Voicemail page
4   In the Voicemail Assistant Telephone Number field enter the internal or external number to which callers are to be transferred.
5   Select Update or Apply when ready.

General	Details	Telephony	DND	Capability	Tunes	Speed Dials	Voicemail	Licenses

User Voicemail (Alice Barker)

Voicemail Enabled	☑
Voicemail Access Code	
Voicemail Assistant Telephone Number	2001

If a caller elects to be transferred to the "Assistant" and the call is not answered the caller will be forwarded to the Assistant's voicemail.

This feature can also be set by the User via a PCS 60, PCS 50, PCS 580/570/560 or 410/400.

Please note: Ensure that the relevant Voicemail Port has been configured with a Dial Plan otherwise the voicemail server will not be able to dial the digits entered in the Assistant Telephone field.

Accessing Voicemail

Users will be informed that they have a new message by one of the following methods:

Via an analogue handset - broken dial tone will be heard

Via Caller Display on an analogue handset
- message waiting LED
- "message waiting" message
- handset will ring once and the number of new messages displayed

Via the PCS 580 – the SpliceCom LED will flash

Via PCS 560/570 - the New Message notice will appear within the default idle page and the SpliceCom LED will flash

Via PCS 410/400 – the New Messages icon and stylus LED will flash

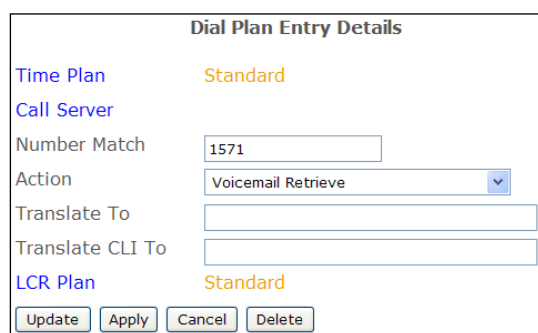Via PCS 100 – the Message Waiting Indicator LED with flash

Via PCS 60 – the Messages icon will flash

Via PCS 50 – the New Messages icon with flash

Users can access their messages via a Dial Plan entry, via Voicemail Call Back or via their PCS (please refer to the relevant PCS User manual for further details).

If a Dial Plan entry is to be used the following gives an example for voicemail retrieval:

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	1571
Action	Voicemail Retrieve
Translate To	
Translate CLI To	
LCR Plan	Standard

[Update] [Apply] [Cancel] [Delete]

The Translate To field is left blank so that the short code can be used by all Users and the extension on which the short code is dialled determines which User's voicemail is accessed.
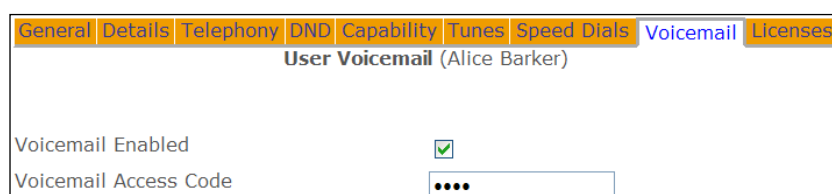
(This is also the default Dial Plan entry for voicemail access in the Standard Dial Plan.)

When accessing their voicemail, if the User has not been assigned a MessageBox licence, they will receive a message indicating that this facility is not available.

Securing access to a User's messages

If a Dial Plan entry is to be used to access voicemail this access can be protected by an access code which can be configured as follows:

1 In Manager select Users
2 From the Users list select the User required
3 Select the Voicemail page
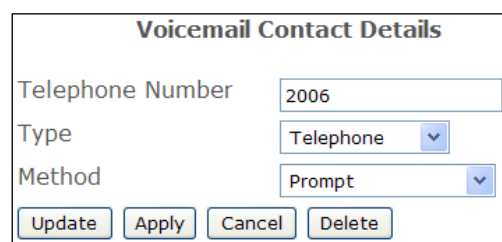4 In the Voicemail Access Code field enter a 1-8 digit code
5 Select Apply

General	Details	Telephony	DND	Capability	Tunes	Speed Dials	**Voicemail**	Licenses

User Voicemail (Alice Barker)

Voicemail Enabled	☑
Voicemail Access Code	••••

6 Select the Add Voicemail Contact button (at the bottom of the page)
7 In the Telephone Number field enter the User's extension number
8 From the Type list box select Telephone
9 From the Method list box select Prompt
10 Select Update or Apply when ready.

Voicemail Contact Details
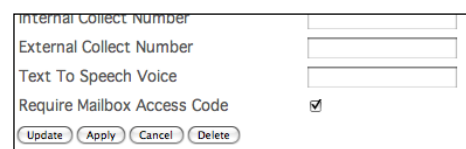
Telephone Number	2006
Type	Telephone
Method	Prompt

[Update] [Apply] [Cancel] [Delete]

Secure all Users' messages

To assign this feature to all Users the Require Mailbox Access Code field can be enabled within the relevant Voicemail Port. Once this feature has been enabled all Users whose voicemail is hosted by the Call Server specified in the Voicemail Port will be prompted to enter their Voicemail Access Code when accessing their voicemail via a dial plan entry.

Internal Collect Number	
External Collect Number	
Text To Speech Voice	
Require Mailbox Access Code	☑

[Update] [Apply] [Cancel] [Delete]

Using Voicemail Call Back

The Voicemail Call Back facility can be configured to automatically ring the User when a new message has been received and on answer deliver the new message.

1   In Manager, select Users
2   Select the User required
3   Select the Voicemail page
4   Select the Add Voicemail Contact button (at the bottom of the page)
5   In the Telephone Number field enter the User's extension number.
6   From the Type list box select Telephone
7   From the Method list box select Deliver
8   Select Update or Apply when ready.

Voicemail Contact Details

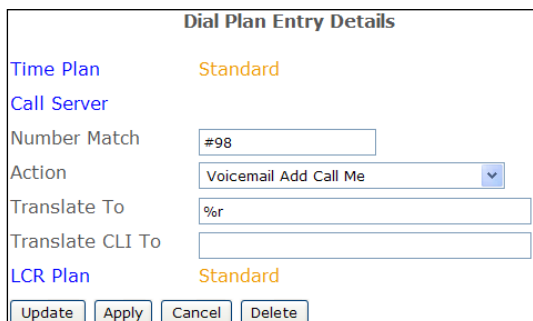|	
Telephone Number	2006
Type	Telephone
Method	Deliver

Update   Apply   Cancel   Delete

Voicemail will attempt to deliver a new message immediately the message is received. If this is not successful this will be repeated after 3 minutes, 10 minutes, 30 minutes, 1 hour and every 3 hours until the call back has been answered.

Please note: Ensure that the relevant voicemail Port has been configured with a Dial Plan otherwise the voicemail server will not be able to dial the digits entered in the Telephone Number field.

A Voicemail Contact entry as describe above can also be created via a Dial Plan entry similar to the following example:

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	#98
Action	Voicemail Add Call Me
Translate To	%r
Translate CLI To	
LCR Plan	Standard

Update   Apply   Cancel   Delete

By dialling, for example, #982006#, the above example Voicemail Contact entry would be created for the User on whose extension the digits where dialled.

A Voicemail Contact entry created to use the Voicemail Call Back facility can also be removed via a Dial Plan entry similar to the following examples:

Dial Plan Entry Details

Time Plan   Standard
Call Server
Number Match   ##98
Action   Voicemail Remove Call Me
Translate To   %r
Translate CLI To
LCR Plan   Standard
[Update] [Apply] [Cancel] [Delete]

Dial Plan Entry Details

Time Plan   Standard
Call Server
Number Match   ##99
Action   Voicemail Remove All Call Me
Translate To
Translate CLI To
LCR Plan   Standard
[Update] [Apply] [Cancel] [Delete]

By dialling, for example, ##982006# the Voicemail Contact entry created above would be removed.

Alternatively, the above example Dial Plan entry can be used to remove all Voicemail Contacts created to provide the Voicemail Call Back facility.

Voicemail Call Back can be configured to call the User on another extension or external number, please refer to Remote Access to Voicemail on page 202 for further details.

Listening to Messages

On accessing their voicemail via a Dial Plan entry or via Voicemail Call Back the User will be informed how many new messages they have. Each new message will be played followed by the time of the call. If the User has any old messages these will then be played. (Please refer to the relevant PCS User manual for details on listening to messages via a PCS 580/570/560/410/400/100/60/50.)

Messages are stored until listened to and are then deleted after 7 days unless manually saved. A message is marked as old after listening to the first 10 seconds.

To assist with the management of messages the following options are available:

1   Play the Help message, then return to your previous position

2   Skip forward by 2 seconds

3   Delete the message you are currently listening to

4   Play previous message

5   Forward the message you are currently listening to
     1   Forward message to another voicemail box with a message attached
     2   Forward message to another voicemail box
     3   Forward message to email
     0   To cancel

6   Play next message

7   Save the message you are currently listening to

8   Skip back by 2 seconds

9   Play the source of the call for the message you are currently listening to. (This feature will only be available if the caller's CLI has been presented with the call.)

> 0   Call back the source. (This feature will only be available if the caller's CLI has been presented with the call and the relevant Voicemail Port has been configured with a Dial Plan to support this feature.)
>
> *   Settings

At the end of all messages the User will be played "For help, please press 1" and the above list of options will be detailed.

Forwarding Voicemail messages

A User can forward a message, to another colleague with or without an annotation or to an email account, by pressing 5 after listening to the message.

Forward a message to colleague(s) with a message attached

1   After listening to the message, press 5
2   Press 1
3   When prompted, record the message to be attached and finish with a hash (#)
4   When prompted, enter the extension number of the User to receive the message followed by a hash (#), enter the next extension number followed by a hash (#), etc. Finish with a hash (#), eg 2001## or 2001#2002##.
5   The next message or "For help, please press 1" will be played.

Forward a message to a colleague(s) without a message attached

1   After listening to the message, press 5
2   Press 2
3   When prompted, enter the extension number of the User to receive the message followed by a hash (#), enter the next extension number followed by a hash (#), etc. Finish with a hash (#), eg 2001## or 2001#2002##.
4   The next message or "For help, please press 1" will be played.

Forward a message to an email account

1   After listening to the message, press 5
2   Press 3
3   The next message or "For help, please press 1" will be played.

Please refer to the Setting up Voicemail Email section from page 206 for further details.

Voicemail Settings

After accessing their voicemail a User can press * and will be given the following options:

1   Repeat the options
2   Record a greeting
3   Select Forwarding
4   Add a voice tag

5   Change your access code
6   Record extra greetings

Recording a greeting

A User can record a daily greeting as follows:

1   On accessing voicemail press *
2   Press 2
3   The User's current daily greeting will be played
4   "Speak greeting after the tone followed by hash or silence" will be played
5   Record the new greeting and press # when finished
6   The new greeting will be played back
7   The following options will be listed:
    1   Repeat the options
    2   Re-record the greeting
    3   Delete the greeting (Restore the previous greeting or return to the default message.)
    7   or Hang up to save the greeting (and return to the previous menu)
    0   Discard the greeting (Do not save new greeting and return to previous menu.)

Please note:
- If a User has recorded a permanent greeting the User's daily greeting will be deleted at midnight and their permanent greeting will be used until a new daily greeting is recorded.
- If a User has selected an Out of Office message the relevant greeting will be played
- A User can also change his greeting via a Dial Plan entry using the VoicemailGreeting Action. Please refer to page 199 for further details.

Setting Forwarding via voicemail

Users can set their Follow Me facility via voicemail; this is particularly useful when Users are out of the office and dialling in via, for example, their mobiles. (Please note that this feature will set the User's Follow Me list box to Personal therefore only calls to the User's extension number and DDI are forwarded.)

Under the Details page of the User's configuration form a User's home number, mobile number and a further two numbers can be stored. Voicemail can use these numbers for forwarding:

1   On accessing voicemail press *
2   Press 3
3   The following options will be listed:
    1   Repeat the options
    2   Cancel Forwarding
    3   Forward to Home
    4   Forward to Mobile
    5   Forward to Spare1
    6   Forward to Spare2
    7   Forward to here (the CLI of phone being used to access voicemail will be entered as the Follow Me number)
    0   Return to settings

Assigning a Voice Tag to incoming CLI

When a call is received by voicemail the CLI of the caller, if presented, is recorded. To assist with the management of voicemail messages a recorded name can be assigned to this CLI so that voicemail will inform the User of the name of the caller before playing the message.

1   On accessing voicemail press *
2   Press 4
3   "Please speak name for <number> followed by hash or silence" will be played
4   The following options will be listed:
      1   Repeat the options
      2   Re-record the tag
      3   Delete the tag
      7   Save the tag
      0   Return to the previous menu

Changing the Voicemail Access Code

Each User can have an access code to enable remote access to voicemail as described from page 202. This code can be set as follows:

1   On accessing voicemail press *
2   Press 5
3   "Enter a new Access Code followed by hash" message will be placed
4   Enter the new code required eg 4567#
5   "Your new Access Code has been set" will be played and you will be returned to the previous menu.

Recording Extra Greetings

As well as the daily greeting a User can record 5 other greetings depending on their current state, whether on holiday, at lunch and so on. The full list is given below and a description of each greeting is described in the Recording a Greeting section below.

1   On accessing voicemail press *
2   Press 6
3   The following options will be listed:
      1   Repeat the options
      2   Record a normal daily greeting
      3   Record a permanent daily greeting
      4   Record an In Meeting greeting
      5   Record an At Lunch greeting
      6   Record an On Holiday greeting
      7   Record an Off Site greeting
      0   Return to Settings

Recording a Greeting

Each User has the facility to record up to 6 voicemail greetings.

Firstly, a User can record a daily greeting that can be used to inform a caller of a User's whereabouts on a particular day, for example – "You have reached Alice Barker's voicemail on Tuesday, 4 July. I am currently in a meeting until 11 o'clock. Please leave a message and I will call you on my return."

The permanent greeting can be used to give a message that can be used on any day, for example – "You have reached Alice Barker's voicemail. Sorry I am unable to take your call. Please leave a message and I will get back to you as soon as possible." If a permanent greeting is recorded the daily greeting is deleted at 1 am and the permanent greeting is then used until a new daily greeting is recorded.

The remaining four greetings are used with the Out of Office facility described from page 97. (Please note that the Permanent greeting will be used until a new greeting is recorded.)

The In Meeting greeting will be used when a User has selected the In Meeting Out of Office message on their extension.

The At Lunch greeting will be used when a User has selected the At Lunch Out of Office message on their extension.

The On Holiday greeting will be used when a User has selected the On Holiday Out of Office message on their extension.

The Off Site greeting will be used when a User has selected the Off Site Out of Office message on their extension.
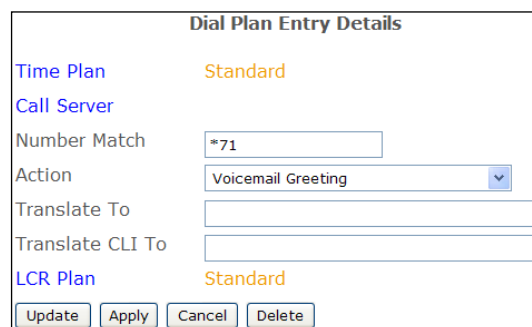
Recording a Daily Greeting

A User can record their daily greeting via 3 methods:

1   Via Settings
   a   On accessing voicemail press *
   b   Press 2
   c   The User's current daily greeting will be played
   d   "Speak greeting after the tone followed by hash or silence" will be played
   e   Record the new greeting and press hash (#) when finished
   f   The new greeting will be played back
   g   The following options can then be used to save or re-record the greeting:
      1   Repeat the options
      2   Re-record the greeting
      3   Delete the greeting (Restore the previous greeting or return to the default message.)
      7   or Hang up to save the greeting (and return to the previous menu)
      0   Discard the greeting (Do not save the new greeting and return to the previous menu)

2   Via Settings and Extra Greetings
   a   On accessing voicemail press *
   b   Press 6
   c   Press 2
   d   Follow the instructions above from step 3

3   Via a Dial Plan entry

A daily greeting can be recorded via a Dial Plan entry similar to This example.

The Translate To field contains the User's extension number or name. If left blank the short code will access the greeting for the User on whose extension the short code was dialled

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	*71
Action	Voicemail Greeting
Translate To	
Translate CLI To	
LCR Plan	Standard

Update   Apply   Cancel   Delete

a   On dialling the short code the current greeting will be played.
b   "Speak greeting after the tone" will be played
c   Record the new greeting and press # when finished
d   The new greeting will be played back
e   The following options can then be used to save or re-record the greeting:
  1   Repeat the options
  2   Re-record the greeting
  3   Delete the greeting (Restore the previous greeting or return to the default message.)
  7   or Hang up to save the greeting (and return to the previous menu)
  0   Discard the greeting (do not save the new greeting and return to the previous menu)

Please note: If a User has recorded a permanent greeting the User's daily greeting will be deleted at midnight and their permanent greeting will be used until a new daily greeting is recorded.
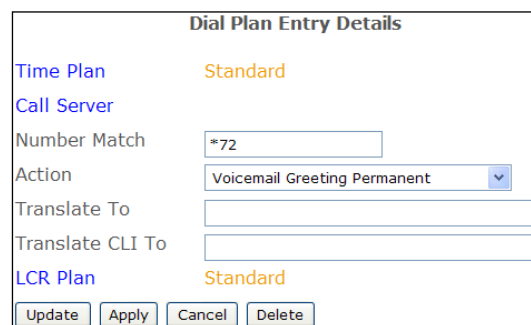
Recording a Permanent Greeting

A User can record a permanent greeting via 2 methods.

1   Via Settings and Extra Greetings
  a   On accessing voicemail press *
  b   Press 6
  c   Press 3
  d   The User's current permanent greeting will be played
  e   "Speak greeting after the tone followed by hash or silence" will be played
  f   Record the new greeting and press hash (#) when finished
  g   The new greeting will be played back
  h   The following options can then be used to save or re-record the greeting:
    1   Repeat the options
    2   Re-record the greeting
    3   Delete the greeting (Restore the previous greeting or return to the default message.)
    7   or Hang up to save the greeting (and return to the previous menu)
    0   Discard the greeting (do not save the new greeting and return to the previous menu)

2   Via a Dial Plan entry

A Permanent greeting can be recorded via a Dial Plan entry similar to this example.

The Translate To field contains the User's extension number or name. If left blank the short code will access the greeting for the User on whose extension the short code was dialled.

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	*72
Action	Voicemail Greeting Permanent
Translate To	
Translate CLI To	
LCR Plan	Standard

Update Apply Cancel Delete

a   On dialling the short code the current greeting will be played.
b   "Speak greeting after the tone" will be played
c   Record the new greeting and press hash (#) when finished
d   The new greeting will be played back
e   The following options can then be used to save or re-record the greeting:
      1   Repeat the options
      2   Re-record the greeting
      3   Delete the greeting (Restore the previous greeting or return to the default message.)
      7   or Hang up to save the greeting (and return to the previous menu)
      0   Discard the greeting (do not save the new greeting and return to the previous menu)

Recording an Out of Office greeting

When a User has set an Out of Office message, as described in the Using an Out of Office Message section from page 97, and a caller is passed to voicemail the relevant voicemail greeting is played. These greetings can be recorded as follows:

1   On accessing voicemail press *
2   Press 6
3   Press
      4   to record an In Meeting greeting
      5   to record an At Lunch greeting
      6   to record an On Holiday greeting
      7   to record an Off Site greeting
4   The User's current permanent greeting will be played
5   "Speak greeting after the tone followed by hash or silence" will be played
6   Record the new greeting and press hash (#) when finished
7   The new greeting will be played back
8   The following options can then be used to save or re-record the greeting:
      1   Repeat the options
      2   Re-record the greeting
      3   Delete the greeting (Restore the previous greeting or return to the default message.)
      7   or Hang up to save the greeting (and return to the previous menu)
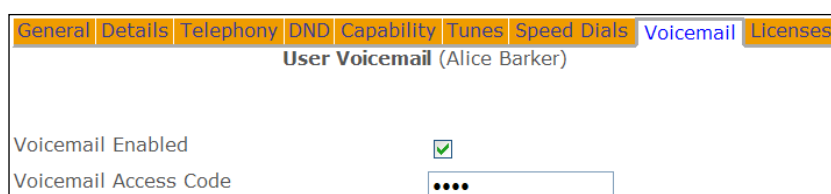   0   Discard the greeting (do not save the new greeting and return to the previous menu)

Remote Access to Voicemail

Users may wish to access their messages from their mobiles, when working at home, from another extension other than their own when, for example, they are working in another office or have an assistant who monitors their voicemail, etc. A Voicemail Access Code must be configured for the User.

Setting a User's Voicemail Access Code

To allow remote access to a User's voicemail an Access Code must be configured as follows:

1. In Manager select Users
2. From the Users list select the User required
3. Select the Voicemail page
4. In the Voicemail Access Code field enter a 1-8 digit code
5. Select Update or Apply when ready.

General	Details	Telephony	DND	Capability	Tunes	Speed Dials	Voicemail	Licenses

User Voicemail (Alice Barker)

Voicemail Enabled	☑
Voicemail Access Code	••••

Users can set their own Voicemail Access Code via Settings. Please refer to page 197 for further details.
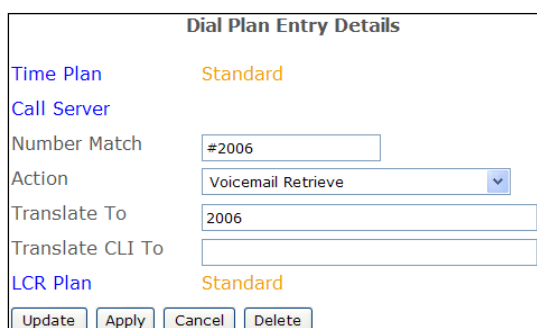
Default Remote Access

All Users can access their voicemail messages from any location by dialling an asterisk (*) while listening to their own greeting message. The User will be prompted for their Access Code, this number should be entered followed by a hash (#).

However if direct access is required the following functionality can be configured.
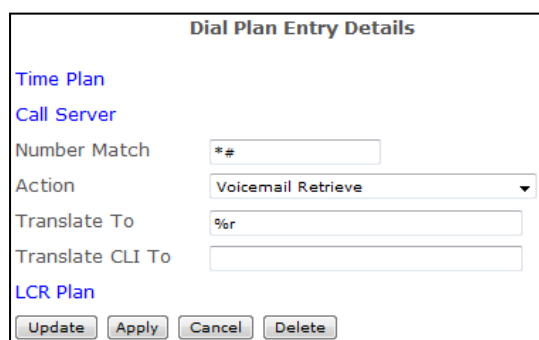
Accessing Voicemail from another extension

A User may wish to access voicemail from other extensions when away from their desk or have an assistant responsible for managing voicemail messages. A Dial Plan entry can be created to enable this facility similar to the following example.

A Voicemail Access Code must be configured for each User wishing to access their voicemail from another extension. Please refer to page 203 for details on how to configure an Access Code.

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	#2006
Action	Voicemail Retrieve
Translate To	2006
Translate CLI To	
LCR Plan	Standard

[Update] [Apply] [Cancel] [Delete]

The Translate To field must contain the User's extension number. This entry must be created within the Dial Plan used by the extensions on which the short code will be dialled.

or

Dial Plan Entry Details

Time Plan

Call Server

Number Match	*#
Action	Voicemail Retrieve ▼
Translate To	%r
Translate CLI To	

LCR Plan

[Update] [Apply] [Cancel] [Delete]

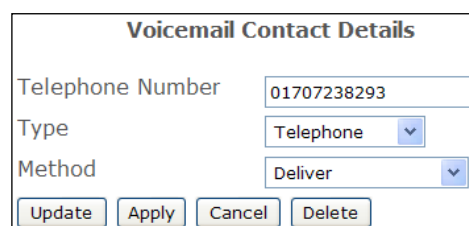A general Short Code can be created for every User.

The User would enter, for example, *#2006 and access voicemail for extension 2006.

In both cases the Voicemail Access Code for the relevant extension will be required which should be entered when prompted followed by a hash (#).

Using Voicemail Call Back

To automatically deliver new messages to a mobile or home telephone number or to another extension, eg an assistant's extension, Voicemail Call Back can be configured to provide this facility.

1 In Manager, select Users
2 Select the User required
3 Select the Voicemail page
4 Select the Add Voicemail Contact button (at the bottom of the page)
5 In the Telephone Number field enter the external telephone or internal extension number required
6 From the Type list box select Telephone
7 From the Method list box select Deliver
8 Select Update or Apply when ready.

Voicemail Contact Details

Telephone Number	01707238293
Type	Telephone ▼
Method	Deliver ▼

[Update] [Apply] [Cancel] [Delete]

Voicemail will attempt to deliver a new message immediately the message is received. If this is not successful this will be repeated after 3 minutes, 10 minutes, 30 minutes, 1 hour and every 3 hours until the call back has been answered. If during this time a User has accessed the voicemail box and listened to the new message the call back will be cancelled.

Please note:
- If the Deliver New Method is selected the call back will operate as described above however the message will not be preceded by the usual statement indicating the number of new messages received. This is useful if a voicemail message is to be announced via a paging port.
- Ensure that the relevant Voicemail Port has been configured with a Dial Plan otherwise the voicemail server will not be able to dial the digits entered in the Telephone Number field.

A Voicemail Contact entry as described above can also be created via a Dial Plan entry similar to the following example:

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	#98
Action	Voicemail Add Call Me
Translate To	%r
Translate CLI To	
LCR Plan	Standard

[Update] [Apply] [Cancel] [Delete]

By dialling, for example, #9801707238293#, the above example Voicemail Contact entry would be created for the User on whose extension the digits where dialled.

A Voicemail Contact entry created to use the Voicemail Call Back facility can also be removed via a Dial Plan entry similar to the following examples:

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	##98
Action	Voicemail Remove Call Me
Translate To	%r
Translate CLI To	
LCR Plan	Standard

[Update] [Apply] [Cancel] [Delete]

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	##99
Action	Voicemail Remove All Call Me
Translate To	
Translate CLI To	
LCR Plan	Standard

[Update] [Apply] [Cancel] [Delete]

By dialling, for example, ##9801707238293# the Voicemail Contact entry created above would be removed.

Alternatively, the above example Dial Plan entry can be used to remove all Voicemail Contacts created to provide the Voicemail Call Back facility.

Accessing Voicemail externally

A DDI number can be linked directly to voicemail running on the Call Server so that User's can have direct access to their messages from an external location, eg their mobile, while at home etc.

A Voicemail Access Code must be configured for each User wishing to access their voicemail externally. Please refer to page 203 for details on how to configure an Access Code.

To use this facility a DDI Call Plan entry, similar to this example, can be created.

On dialling the DDI number a User will be prompted for their Account Code, at this point the User should enter their extension number followed by a hash (#). The User will then be prompted for their Access Code, this number should be entered followed by a hash (#).

DDI Plan Entry Details

Order	1
Time Plan	Standard
Number Match	402050
Bearer Capability	Voice
Caller ID	
Translate To	!CollectVoicemail
Locale	

[Update] [Apply] [Cancel] [Delete]
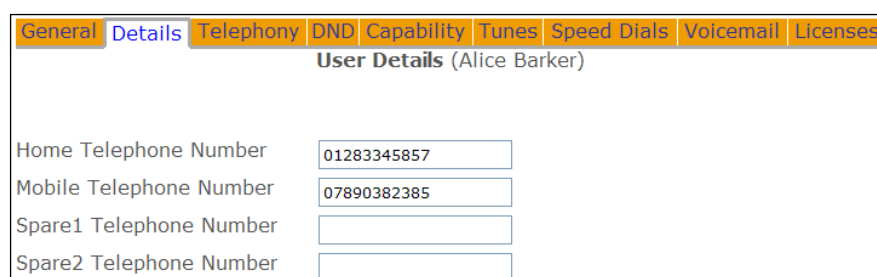
Dialling from a trusted location

When a User dials a DDI number to access voicemail, as described above, from a location where the incoming CLI matches a number entered in their Home Telephone Number, Mobile Telephone Number,

Spare 1 Telephone Number or Spare 2 Telephone Number fields, found within the Details page of the User's configuration form, the User will not be prompted for their Account Code.
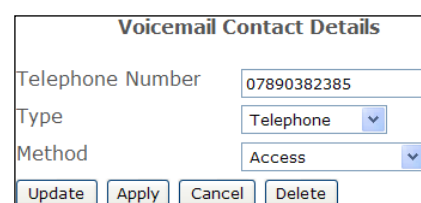
General	Details	Telephony	DND	Capability	Tunes	Speed Dials	Voicemail	Licenses

User Details (Alice Barker)

Home Telephone Number	01283345857
Mobile Telephone Number	07890382385
Spare1 Telephone Number	
Spare2 Telephone Number	

If a User does not wish to enter the Access Code when ringing from their mobile or home number, for example, as these are trusted locations, this number can be entered as a Voicemail Contact as follows:

1   In Manager select Users
2   From the Users list select the User required
3   Select the Voicemail page
4   Select the Add Voicemail Contact button (at the bottom of the page)
5   In the Telephone Number field enter the number to be matched with the incoming CLI, for example, the User's mobile number.
6   From the Type list box and select Telephone
7   From the Method list box and select Access
8   Select Update or Apply when ready.
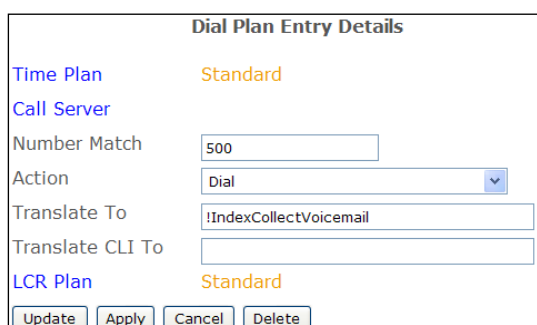
Voicemail Contact Details

Telephone Number	07890382385
Type	Telephone
Method	Access

Update   Apply   Cancel   Delete

Both of these features can be configured for the same number or independently, in other words, one feature is not dependent on the other.

Using voicemail from an INDeX

The following example Dial Plan entries can be used to allow access to the appropriate functions in the User's **max**imiser voicemail box from the appropriate soft buttons on the DT and 20 Series INDeX digital handsets.

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	500
Action	Dial
Translate To	!IndexCollectVoicemail
Translate CLI To	
LCR Plan	Standard

Update   Apply   Cancel   Delete

This example Dial Plan entry will allow Users access to their voicemail. Programme the Listen button to dial this entry.

Dial Plan Entry Details	
Time Plan	Standard
Call Server	
Number Match	502
Action	Dial
Translate To	!IndexGreeting
Translate CLI To	
LCR Plan	Standard
Update Apply Cancel Delete	

This example Dial Plan entry will allow Users to recording their daily greeting. Programme the Temp Msg button to dial this entry.

Dial Plan Entry Details	
Time Plan	Standard
Call Server	
Number Match	501
Action	Dial
Translate To	!IndexLeaveMessage
Translate CLI To	
LCR Plan	Standard
Update Apply Cancel Delete	

This example Dial Plan entry will allow Users to leave a message for another User. Programme the Message button to dial this entry.

Dial Plan Entry Details	
Time Plan	Standard
Call Server	
Number Match	503
Action	Dial
Translate To	!IndexModify
Translate CLI To	
LCR Plan	Standard
Update Apply Cancel Delete	

This example Dial Plan entry will allow access to change their setting, such as their voicemail access code, for example. Programme the Modify button to dial this entry.

Dial Plan Entry Details	
Time Plan	Standard
Call Server	
Number Match	504
Action	Dial
Translate To	!IndexGreetingPermanent
Translate CLI To	
LCR Plan	Standard
Update Apply Cancel Delete	

This example Dial Plan entry will allow Users to recording their permanent greeting.

Setting up Voicemail Email
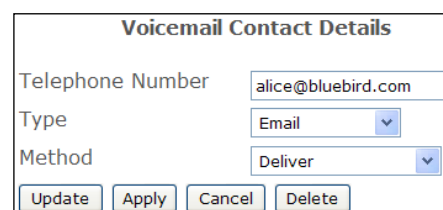
A User may wish to be notified by email that they have received a new message, or they may wish to have their new messages sent to an email account, or they may wish to manually forward a message to email as and when they require. In order to allow all this functionality on the **max**imiser system Voicemail Email must be configured as follows.

1   In Manager select Voicemail Ports
2   Select the required Voicemail Port

3   Within the Email Smart Host field enter the IP address of the SMTP mail server
4   Within the Email Source Address enter the name of the account that will be displayed in the From field of an email.
5   Select Update or Apply when ready.

Configuring a User to utilize this facility

1   In Manager select Users
2   Select the User required
3   Select the Voicemail page
4   Select the Add Voicemail Contact button (at the bottom of the page)
5   In the Telephone Number field enter the email address to receive the messages.
6   From the Type list box select Email
7   From the Method list box select the delivery method required:
   a   Alert – inform the User of a new voicemail message via an email message
   b   Deliver – send the new voicemail message in an email message. The voicemail message will be marked as old on the voicemail server and deleted after 7 days.
   c   Copy – send the new voicemail message in an email message. The voicemail message will still be considered a new message on the voicemail server.
   d   Deliver and Delete – send the new voicemail message in an email message. Immediately delete the voicemail message from the voicemail server.
8   Blank – User can manually send a voicemail message to an email address by pressing 5 and then 3 after listening to the message or via their PCS.
9   Select Update or Apply when ready.
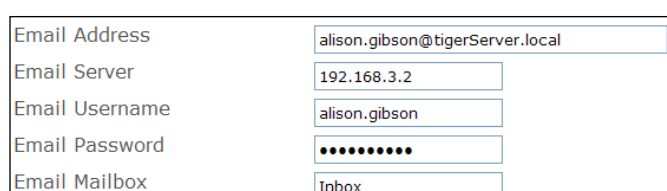
IMAP Unified Messaging

A User's voicemail messages can be sent to any email server supporting Internet Messaging Application Protocol (IMAP) including Lotus Notes, Novell GroupWise, AppleMail and Microsoft Exchange. This facility allows synchronisation between voicemail and email, whereby if a voicemail message is deleted via voicemail the corresponding email will be deleted and visa versa.

1   In Manager select Users
2   Select the User required
3   Select the Voicemail page
4   In the Email Address field enter the email address to receive the User's messages
5   In the Email Server field enter the IP address of the email server supporting this facility
6   In the Email Username field enter the name for the account on the email server.
7   In the Email Password field enter the password for the above account
8   In the Email Mailbox field enter the folder within the account above to receive the messages eg Inbox
9   Select Update or Apply when ready.
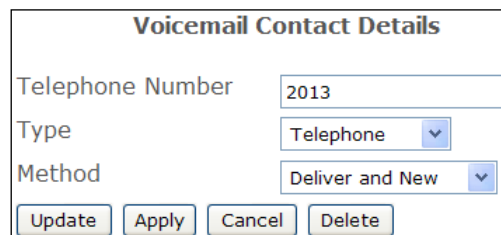
Setting up SMS Alerts

If a User wishes to be notified of a new voicemail message via an SMS alert this can also be configured within the Voicemail page of the User's configuration form.

1   In Manager, select Users
2   Select the User required
3   Select the Voicemail page
4   Select the Add Voicemail Contact button (at the bottom of the page)
5   In the Telephone Number field enter the email address from the SMS provider.
6   From the Type list box select SMS.
7   From the Method list box select Alert.
8   Select Update or Apply when ready.

Using voicemail with a Paging Port

Voicemail can be used to play messages over a tannoy or PA system connected to the system. In a busy environment this means that several people can be leaving messages rather than having to wait for the paging extension to become free. For details on how to configure a paging port please refer to page 78.

1   Create a new User with an extension number not currently in use
2   Enable voicemail for this User.
3   Within the Voicemail page click on the Add Voicemail Contact button
4   In the Telephone Number field enter the extension number for User assigned to the port to which the PA system is connected.
5   From the Type list box select Telephone
6   From the Method list box select Deliver and New
7   Select Update or Apply when ready.
8   You may wish to turn on Do Not Disturb for this User.

To play a message over the PA system ring the User created above and leave a message. Voicemail will then automatically ring the extension number entered in the Telephone Number field and play the message.

Voicemail for a Department

Enabling voicemail

In order to provide a Department with voicemail functionality a Message Box Licence must be assigned to that Department.

1   In Manager select Departments
2   From the Departments list select the Department required
3   Select the Licences page
4   Tick the Message Box Licence field
5   Select Update or Apply when ready.

splicecom maximiser

General	Distribution	Telephony	Voicemail	Licenses

Department Licenses (Sales)

Message Box License   ☑

[Update] [Apply] [Cancel] [Delete]

If there are insufficient licences available these settings will not be saved. A list of which Departments have been assigned a MessageBox licence can be viewed in Manager by selecting Utilities and then Mailbox Licences.

Once a MessageBox licence has been assigned, a Department will have use of the following facilities:

- Listen to messages (as described in the Accessing voicemail section from page 214).
- Users will be able to transfer calls to the Department's voicemail (as described in the Transfer to Voicemail section from page 211).
- Call recording (as described in the Call Recording section from page 232).

Further voicemail functionality can be provided to a Department by configuring the Voicemail Enabled option as follows:

1   In Manager select Departments
2   From the Departments list select the Department required
3   Select the Voicemail page
4   Tick the Voicemail Enabled field
5   Select Apply

General	Distribution	Telephony	Voicemail	Licenses

Department Voicemail (Sales)

Voicemail Enabled   ☑

Once this option has been enabled callers to a Department will be transferred to voicemail after the time specified in the Max No Answer Time before Voicemail field as described in the Routing Department calls to voicemail section from page 211 or Out of Hours Max No Answer Time before Voicemail field as described in the Routing Department Calls to voicemail out of hours section from page 212.

Specifying the Call Server providing voicemail functionality

The Current Home field within the Department's General page determines which Call Server is providing the Department with the voicemail service.   This is relevant when a system consists of multiple Call Servers and it is necessary to know which Call Server is storing the Department's messages and to ensure the correct Call Server is being used.

Transfer to Voicemail

If a User wishes to be able to dial directly to a Department's voicemail or to transfer a caller to a Department's voicemail this can be done either via their PCS 580/570/560, PCS 410/400, PCS 100, PCS 60 or PCS 50, or via a Dial Plan entry similar to the following examples:

The exclamation mark (!) in the Translate To field = Go to Voicemail.

Routing Department calls to voicemail

If calls to a Department are to be routed to voicemail, to allow callers to leave a message, the Max No Answer Time Before Voicemail option must be set as follows.

1   In Manager select Departments
2   Select the Department required
3   Select the Telephony page
4   Within the Max No Answer Time Before Voicemail field enter the number of seconds that a call will ring for the Department before being passed to voicemail. This time must take into account the amount time of required for the call to ring around the Distribution Group and Alternate Distribution Groups if configured.
5   Select Update or Apply when ready.

General	Distribution	Telephony	Voicemail	Licenses

Department Telephony (Sales)

Max Ring Time Before Announcement	0
Repeat Announcement Time	0
Max Ring Time Before Alternate Distribution	60
Max Ring Time Before Alternate Distribution 2	60
Max Number Of Active Calls Before Busy	10
Wrap Up Time	120
No Answer Time	15
Max No Answer Time Before Voicemail	120

Please note that the Voicemail Enabled field must be ticked to allow callers to be automatically transferred to voicemail after this time (please refer to the Enabling voicemail section from page 209 for further details.)

The caller by default will have 5 minutes to leave a message. This can be changed by entering the number of minutes required in the Max Message Time (Minutes) field within the Department's Voicemail page.

Routing Department Calls to voicemail out of hours

When a Department's Out of Hours mode is set to Out of Hours or Timed and is outside of its operational hours, calls can be passed either directly to voicemail or firstly to the Out of Hours Distribution Group and then if the call is not answered passed to voicemail. The Out of Hours Max No Answer Time Before Voicemail field will determine how long the call will ring before being passed to voicemail.

1. In Manager select Departments
2. From the Departments List select the Department required
3. Select the Telephony page
4. Within the Out of Hours Max No Answer Time Before Voicemail field enter the number of seconds that a call will ring for the Department before being passed to voicemail. If the call is to pass directly to voicemail enter, for example, 1. Alternatively, if the call is pass to the Out of Hours Distribution Group first enter the amount time required for the call to ring around this Group before being passed to voicemail.
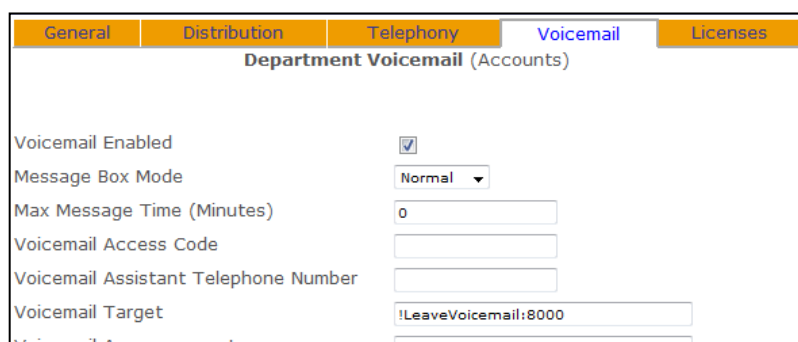5. Select Update or Apply when ready.

General	Distribution	Telephony	Voicemail	Licenses

Department Telephony (Sales)

Max Ring Time Before Announcement	0
Repeat Announcement Time	0
Max Ring Time Before Alternate Distribution	60
Max Ring Time Before Alternate Distribution 2	60
Max Number Of Active Calls Before Busy	10
Wrap Up Time	120
No Answer Time	15
Max No Answer Time Before Voicemail	120
Out Of Hours Max No Answer Time Before Voicemail	60

Please note that the Voicemail Enabled field must be ticked to allow callers to be automatically transferred to voicemail after this time (please refer to the Enabling voicemail section from page 209 for further details.)

Routing Department Calls to an alternative voicemail box

If callers to a Department are to be routed to another Department's voicemail, for example, where a centralised collection point is required configure the following.

1   In Manager select Departments
2   From the Departments list select the Department required
3   Select the Voicemail page
4   In the Voicemail Target field enter *!LeaveVoicemail:extension no* where the extension number is the extension of the Department or User to receive the calls, for example !LeaveVoicemail:8000.
5   Select Update or Apply when ready.

General	Distribution	Telephony	Voicemail	Licenses

Department Voicemail (Accounts)

Voicemail Enabled	☑
Message Box Mode	Normal ▾
Max Message Time (Minutes)	0
Voicemail Access Code	
Voicemail Assistant Telephone Number	
Voicemail Target	!LeaveVoicemail:8000
Voicemail Announcement	

Please note that the Voicemail Enabled field must be ticked to allow callers to be automatically transferred to voicemail (please refer to the Enabling voicemail section from page 209 for further details.)

Routing Department Calls to a Recorded Message

Callers can be played a message rather than be routed to the Department's voicemail box. This can be configured as follows:

1   Format the WAV file to be used as CCITT A-Law 8.000kHz 8bit mono.
2   Using FTP via a browser or Command Prompt log in as Admin on the relevant Call Server as described in the Admin Access to a Call Server section from page 252.

   If you are using a browser, copy the wav file into this area.

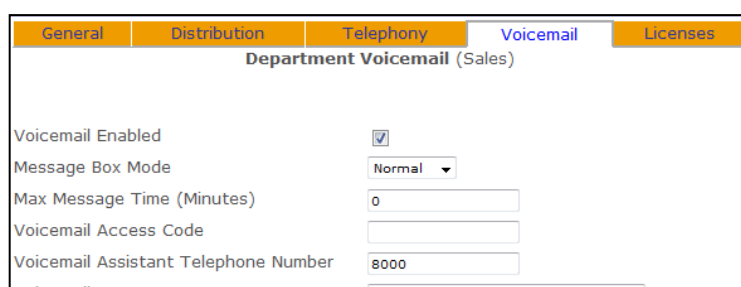   If you are using the Command Prompt:
   a   Place the WAV file in the root, eg C:\
   b   In the Command Prompt enter *put <filename>.wav*, eg put salesmsg.wav, to copy the file from root to the ftp directory.
   c   After receiving the Transmission Complete message enter *Quit*

3   In Manager select Departments
4   From the Departments list select the Department required
5   Select the Voicemail page
6   In the Voicemail Target field enter !Play:<wav file name>, eg !Play:salesmsg.wav
7   Select Update or Apply when ready

Please note that !Play will play the message once and !PlayLoop will repeat the message until the call is ended.

Using the Assistant Telephone Number

Each Department can be configured with an internal or external number to which callers can be transferred if they do not wish to leave a message. The caller can activate the Assistant Telephone number by pressing 0 during or after the Department's greeting. It is recommended that the Department's greeting is amended to inform the callers of this facility.

1. In Manager select Departments
2. From the Departments list select the Department required
3. Select the Voicemail page
4. In the Voicemail Assistant Telephone Number field enter the internal or external number to which callers are to be transferred.
5. Select Update or Apply when ready.

Please note: Ensure that the relevant Voicemail Port has been configured with a Dial Plan otherwise the voicemail server will not be able to dial the digits entered in the Voicemail Assistant Telephone Number field.
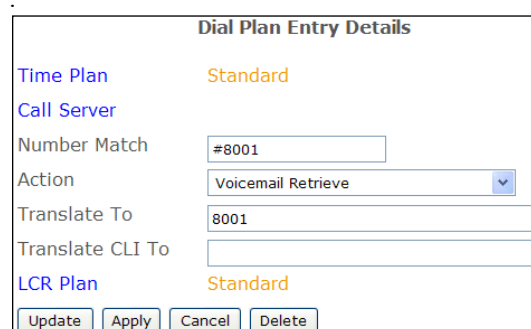
Accessing voicemail

A User can access a Department's messages via a Dial Plan entry, via Voicemail Call Back or via their PCS 580/570/560, 410/400, 100, 60 and 50 (please refer to the Setting Department Voicemail Notification section below for further details).

Accessing Department voicemail via a Dial Plan entry

If a Dial Plan entry is to be used to access voicemail for a Department this diagram gives an example entry that could be created.

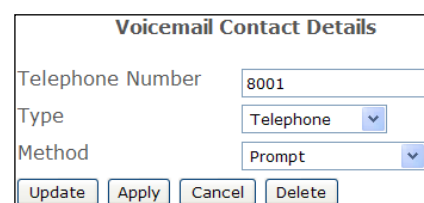The Translate To field must contain the Department's extension number

Please refer to the Working with Dial Plans section from page 114 for further information on creating Dial Plan entries.

Any member of the Department's Distribution Group can use this Dial Plan entry to access the Department's voicemail. If this short code is dialled on an extension that is not a member of the Distribution Group the Department's Access Code will be requested. Please refer to Remote Access to Department Voicemail section on page 221 for further details.

Securing access to a Department's voicemail

If further security is required when using a Dial Plan entry to access a Department's voicemail each member of the Department's Distribution Group can be required to enter the Department's Voicemail Access Code. This can be configured as follows.

1 In Manager select Departments
2 From the Departments list select the Department required
3 Select the Voicemail page
4 In the Voicemail Access Code field enter a 1-8 digit code
5 Select Apply
6 Select the Add Voicemail Contact button
7 In the Telephone Number field enter the Department's extension number
8 From the Type list box select Telephone
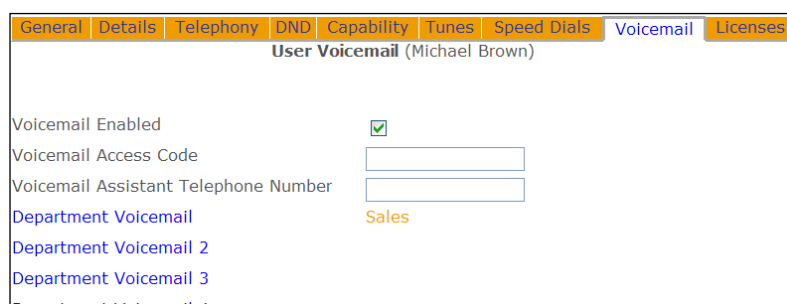9 From the Method list box select Prompt
10 Select Update or Apply when ready.

Please refer to the Setting a Department's Voicemail Access Code section from page 221 for further details.

Setting Department Voicemail Notification

If a User of a PCS 580/570/560, PCS 410/400, PCS 100, PCS 60 or PCS 50 wishes to handle voicemail messages for a Department via their PCS this Department should be entered in one of the User's Department Voicemail fields as follows.

1 In Manager select Users
2 From the Users list select the User required
3 Select the Voicemail page
4 Select the Department Voicemail field
5 From the Select Department list select the Department required
6 Repeat steps 4 and 5 for the Department Voicemail 2, 3, 4, 5, 6, 7 and 8 fields, if required.
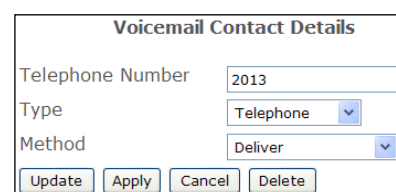7 Select Update or Apply when ready.

Please refer to the relevant PCS User manual for details on how to handle voicemail messages via a PCS. Incoming calls answered and missed by the Department will also be logged in the User's Incoming Calls History.

Please note: a User does not need to be a member of the Department to use this feature.

Using Voicemail Call Back

When a new message is received by a Department the voicemail server can be configured to ring a User and deliver the new message. This can be set as follows:

1   In Manager, select Departments
2   Select the Department required
3   Select the Voicemail page
4   Select the Add Voicemail Contact button
5   In the Telephone Number field enter the User's extension number.
6   From the Type list box select Telephone
7   From the Method list box select Deliver
8   Select Update or Apply when ready.

Voicemail Call Back can also be configured to call an external number, eg a mobile or home telephone number.

Voicemail will attempt to deliver a new message immediately the message is received. If this is not successful this will be repeated after 3 minutes, 10 minutes, 30 minutes, 1 hour and every 3 hours until the call back has been answered.

Please note: Ensure that the relevant Voicemail Port has been configured with a Dial Plan otherwise the voicemail server will not be able to dial the digits entered in the Telephone Number field.

Listening to Messages

On accessing a Department's voicemail box via a Dial Plan or via Voicemail Call Back the User will be informed how many new messages have been received. Each new message will be played followed by the time of the call. If the Department has any old messages these will then be played.

To assist with the management of Department messages the same options as listed on page 196 are available.

(Please refer to the relevant PCS User manual for details on listening to messages via a PCS 580/570/560/410/400/100/60/50.)

Messages are stored until listened to and are then deleted after 7 days unless manually saved. A message is marked as old after listening to the first 10 seconds.

Voicemail Settings

After accessing the Department's voicemail the User can press * and will be given the following options:

    1   Repeat the options
    2   Record a greeting
    3   Select Forwarding
    4   Add a voice tag
    5   Record an initial announcement
    6   Record a repeat announcement

Recording a greeting

    1   On accessing voicemail press *
    2   Press 2
    3   The following options will be listed:
        1   Repeat the options
        2   Record a normal daily greeting
        3   Record an out of hours greeting
        4   Record an In Meeting greeting
        5   Record an At Lunch greeting
        6   Record an On Holiday greeting
        7   Record an Off Site greeting
        0   Return to Settings

Recording a greeting

Recording a daily greeting

A User can record a daily greeting for a Department as follows:
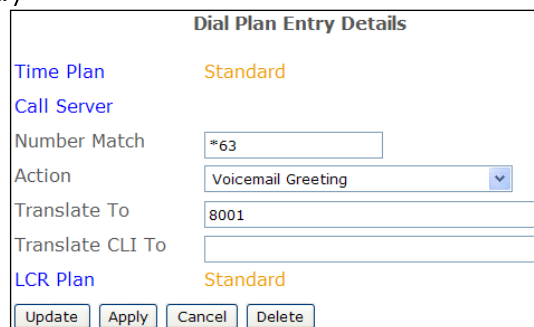
    1   On accessing voicemail for the Department press *
    1   Press 2
    2   Press 2 again to record a normal greeting
    3   The current greeting will be played
    4   "Speak greeting after the tone followed by hash or silence" will then be played
    5   Record the new greeting and press # when finished
    6   The new greeting will be played back
    7   The following options will be listed:
        1   Repeat the options
        2   Re-record the greeting
        3   Delete the greeting (restore the previous greeting or return to the default message.)
        7   or hang up to save the greeting (and return to the previous menu)
        0   Discard the greeting (Do not save new greeting and return to previous menu.)

Please note: If a permanent greeting has been recorded a Department's daily greeting will be deleted at midnight and its permanent greeting will be used until a new daily greeting is recorded.

Recording a daily greeting directly via a Dial Plan entry

A Department's daily greeting can be recorded via Settings as described above. Alternatively, a Dial Plan entry can be used similar to this example.

The Translate To field contains the Department's extension number or name.

Dial Plan Entry Details

Time Plan — Standard
Call Server
Number Match — *63
Action — Voicemail Greeting
Translate To — 8001
Translate CLI To
LCR Plan — Standard

[Update] [Apply] [Cancel] [Delete]

1. On dialling the short code the current greeting will be played.
2. "Speak greeting after the tone" will be played
3. Record the new greeting and press # when finished
4. The new greeting will be played back
5. The following options can then be used to save or re-record the greeting:
   - 2 Re-record the new greeting
   - 3 Delete the new greeting and restore the previous greeting
   - 7 Save the new greeting (or hang up)
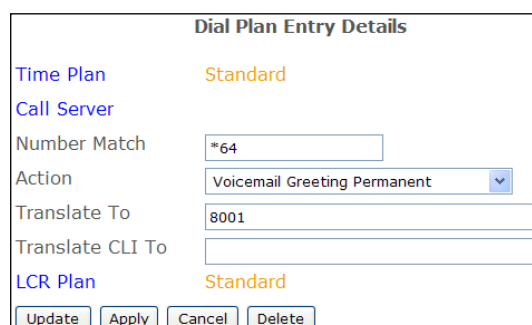   - 0 Do not save new greeting

Please note: If a permanent greeting is recorded a Department's daily greeting will be deleted at midnight and its permanent greeting will be used until a new daily greeting is recorded. Please refer to page 218 for further details.

Recording a permanent greeting

Each Department can record two voicemail greetings. Firstly, the daily greeting which can be recorded via Settings or via a Dial Plan entry as described above and may inform a caller of a Department's status on a particular day, for example – "The Accounts Department is closed today. Please leave a message and we will call you as soon as possible." The second greeting, the permanent greeting, can be used to give a message that can be used on any day, for example – "Thank you for calling the Accounts Department. Sorry we are unable to take your call. Please leave a message and we will get back to you as soon as possible." If a permanent greeting is recorded the daily greeting is deleted at midnight and the permanent greeting is then used until a new daily greeting is recorded.

To create a Permanent greeting a Dial Plan entry similar to this example can be used.

The Translate To field contains the Department's extension number or name

Dial Plan Entry Details

Time Plan — Standard
Call Server
Number Match — *64
Action — Voicemail Greeting Permanent
Translate To — 8001
Translate CLI To
LCR Plan — Standard

[Update] [Apply] [Cancel] [Delete]

1. On dialling the short code the current greeting will be played.
2. "Speak greeting after the tone" will be played
3. Record the new greeting and press # when finished
4. The new greeting will be played back
5. The following options can then be used to save or re-record the greeting:

2   Re-record the new greeting
3   Delete the new greeting and restore the previous greeting
7   Save the new greeting (or hang up)
0   Do not save new greeting

Recording an Out of Hours greeting

The out of hours greeting is played to callers when a Department call has been routed to voicemail when using Out of Hours or Timed Out of Hours Mode (please refer to page 212 for further details). The out of hours greeting can be recorded as follows:

1   On accessing voicemail for the Department press *
2   Press 2
3   Press 3 to record an out of hours greeting
4   The current greeting will be played
5   "Speak greeting after the tone followed by hash or silence" will then be played
6   Record the new greeting and press # when finished
7   The new greeting will be played back
8   The following options will be listed:
    1   Repeat the options
    2   Re-record the greeting
    3   Delete the greeting (restore the previous greeting or return to the default message.)
    7   or hang up to save the greeting (and return to the previous menu)
    0   Discard the greeting (Do not save new greeting and return to previous menu.)

Recording an Out of Office greeting

When an Out of Office message has been configured for a Department, as described in the Using an Out of Office Message section from page 140, and a caller goes to voicemail the relevant voicemail greeting will be played. These can be recorded as follows:

1   On accessing voicemail press *
2   Press 2
3   Press
    4   to record an In Meeting greeting
    5   to record an At Lunch greeting
    6   to record an On Holiday greeting
    7   to record an Off Site greeting
4   The User's current permanent greeting will be played
5   "Speak greeting after the tone followed by hash or silence" will be played
6   Record the new greeting and press hash (#) when finished
7   The new greeting will be played back
8   The following options can then be used to save or re-record the greeting:
    1   Repeat the options
    2   Re-record the greeting
    3   Delete the greeting (Restore the previous greeting or return to the default message.)
    7   or Hang up to save the greeting (and return to the previous menu)
    0   Discard the greeting (do not save the new greeting and return to the previous menu)

Using Announcements

While callers are waiting for their call to be answered comfort announcements can be played. The Max Ring Time before Announcement Time option will specify how long the call will ring before the caller is played the first announcement. The Repeat Announcement Time specifies the amount of time before the second announcement is played and how often this is then played. These announcements are played via the voicemail server and can be configured as follows:

1   In Manager select Departments
2   From the Departments list select the Department required.
3   Select the Telephony page
4   In the Max Ring Time before Announcements field enter the number of seconds required before the first announcement is played
5   In the Repeat Announcement field enter the number of seconds required before the second announcement is played. This will also specify the interval before the second announcement is played again.
6   Select Update or Apply when ready.

General	Distribution	Telephony	Voicemail	Licenses

Department Telephony (Sales)

Max Ring Time Before Announcement	5
Repeat Announcement Time	10

Please note:
- The relevant Voicemail Port must be configured to support this facility, this is done by ticking Queue Message Server.
- Calls will pass to voicemail after the time entered in the Max No Answer Time before Voicemail field. Please refer to page 211 for further details.
- If the Max Number of Active Calls before Busy entry has been exceeded the caller will be given busy.
- The caller will just hear ringing if this facility is not used
- Music on hold will only be played between announcements to external callers.

Recording Announcements

Two announcements can be played to a caller and the default announcements will be used unless re-recorded as follows:

1   Access the Department's voicemail
2   Press *
3   Press 5 to re-record the first announcement
     or
     Press 6 to re-record the second announcement
4   "Speak announcement after the tone then press hash or remain silent" will be played
5   Record the new announcement and press # when finished
6   The new announcement will be played back
7   The following options will be listed:
     1   Repeat the options
     2   Re-record the new announcement

3   Delete the new announcement and restore the previous announcement or return to the default message

7   Save the new announcement and return to the previous menu, or hang up

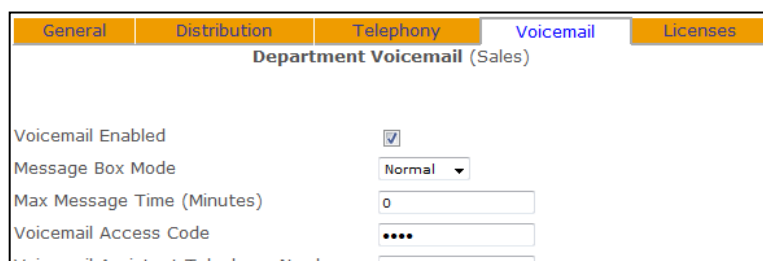0   Do not save new announcement and return to previous menu.

Remote Access to Department voicemail

Users may wish to access Department messages from an extension that is not a member of the Distribution Group, or from a mobile, when working at home etc. A Voicemail Access Code must be configured for the Department.

Setting a Department's Voicemail Access Code

To allow remote access to a Department's voicemail an Access Code must be configured as follows:

1   In Manager select Departments
2   From the Departments list select the Department required
3   Select the Voicemail page
4   In the Voicemail Access Code field enter a 1-8 digit code
5   Select Update or Apply when ready.

General	Distribution	Telephony	Voicemail	Licenses

Department Voicemail (Sales)

Voicemail Enabled	☑
Message Box Mode	Normal ▼
Max Message Time (Minutes)	0
Voicemail Access Code	••••
Voicemail Assistant Telephone Number	••••

Accessing voicemail for a Department externally

A DDI number can be linked directly to voicemail so that User's can have direct access to Department messages from an external location, eg their mobile, while working at home etc. This can be the same number used to allow User's to access their voicemail externally as described on page 205.

If a User does not wish to enter the Access Code when ringing from their mobile or home number, for example, as these are trusted locations, this number can be entered as a Voicemail Contact as follows:

1   In Manager select Departments
2   From the Departments list select the Department required
3   Select the Voicemail page
4   Select the Add Voicemail Contact button
5   In the Telephone Number field enter the number to be matched with the incoming CLI, for example, the User's mobile number.
6   From the Type list box select Telephone
7   From the Method list box select Access
8   Select Update or Apply when ready.

Voicemail Contact Details
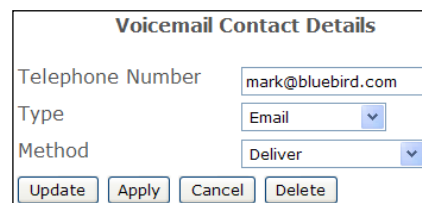
Telephone Number	07890873283
Type	Telephone ▼
Method	Access ▼

| Update | Apply | Cancel | Delete |

Using Voicemail Email

If a User wishes to be informed that a new message has been received for a Department via email the following can be configured. Please note that the Voicemail Email facility must be set up first as detailed on page 206.

1   In Manager select Departments
2   From the Departments List select the Department required
3   Select the Voicemail page
4   Select the Add Voicemail Contact button
5   In the Telephone Number field enter the email address to receive the above messages.
6   From the Type list box select Email
7   From the Method list box select the delivery method required:
    a   Alert – inform the User of a new message via an email message
    b   Deliver – send the new message in an email message. The message will be marked as old on the voicemail server and deleted after 7 days.
    c   Copy – send the new message in an email message. The message will still be considered a new message on the voicemail server.
    d   Deliver and Delete – send the new message in an email message. Immediately delete the message from the voicemail server.
    e   Blank – User can manually send a voicemail message to the email address by pressing 5 and then 3 after listening to the message.
8   Select Update or Apply when ready.

Setting up SMS Alerts

If a User wishes to be notified of a new voicemail message for a Department via an SMS alert this can also be configured within the Voicemail page in the Department's configuration form.

1   In Manager, select Departments
2   From the Departments List select the Department required
3   Select the Voicemail page
4   Select the Add Voicemail Contact button
5   In the Telephone Number field enter the email address provided by the SMS provider.
6   From the Type field select SMS.
7   From the Method field select Alert.
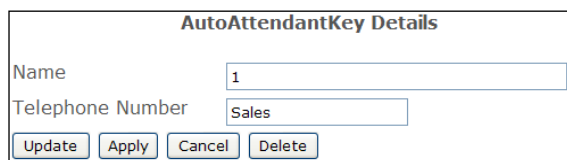8   Select Update or Apply when ready.

Creating an Auto Attendant

The Embedded voicemail provides the ability to configure a single layer Auto Attendant facility so that callers can select the User or Department to which they wish to be transferred.

Firstly, ensure that the relevant Voicemail Port has been configured with a Dial Plan otherwise the voicemail server will not be able to dial the digits selected by the caller and the Auto Attendant Server option has been ticked, then create the Auto Attendant as follows:

1   In Manager select Auto Attendant
2   Select the Add button

3   In the Name field enter the digit to be dialled by the caller to make the required selection. This field can contain numbers 0-9, asterisk (*) and hash (#).

4   In the Telephone Number field enter the extension number, User or Department name to which the caller will be transferred.
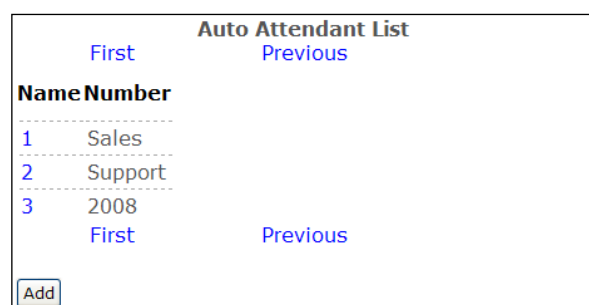
5   Select Update

AutoAttendantKey Details

| Name | 1 |
| Telephone Number | Sales |

Update Apply Cancel Delete

6   Repeat steps 2 to 6 to enter all options to be available to the caller.

Auto Attendant List

First   Previous

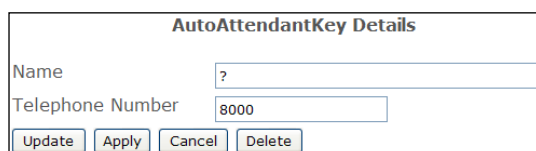| Name	Number
1	Sales
2	Support
3	2008

First   Previous

Add

Using a Timed option

The Auto Attendant can contain an option that will wait 3 seconds for the caller to make a choice. If a choice is not made the call will be automatically transferred to a specified number. This can be configured as follows:

1   In Manager select Auto Attendant

2   Select the Add button

3   In the Name field enter a question mark (?)

4   In the Telephone Number field enter the extension number, User or Department name to which the caller will be transferred.

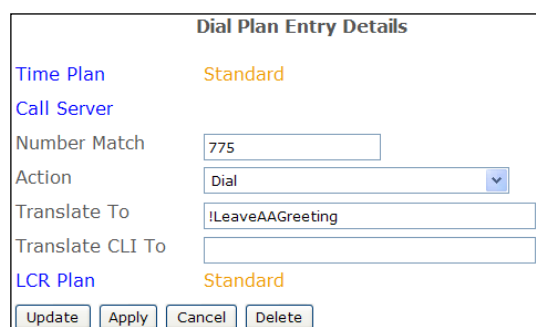5   Select Update or Apply when ready.

AutoAttendantKey Details

| Name | ? |
| Telephone Number | 8000 |

Update Apply Cancel Delete

Record a Greeting Message

To record a greeting to announce the options available to a caller a Dial Plan entry should be created with !LeaveAAGreeting entered in the Translate To field, similar to this example:

Dial Plan Entry Details

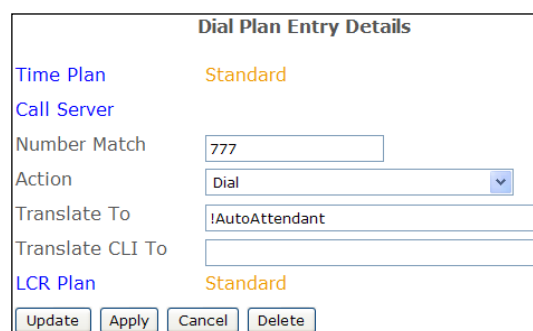| Time Plan | Standard |
Call Server	
Number Match	775
Action	Dial
Translate To	!LeaveAAGreeting
Translate CLI To	
LCR Plan	Standard

Update Apply Cancel Delete

Test the Auto Attendant

To test or use the Auto Attendant internally a Dial Plan entry can be created with !AutoAttendant entered in the Translate To field, similar to this example:

Dial Plan Entry Details

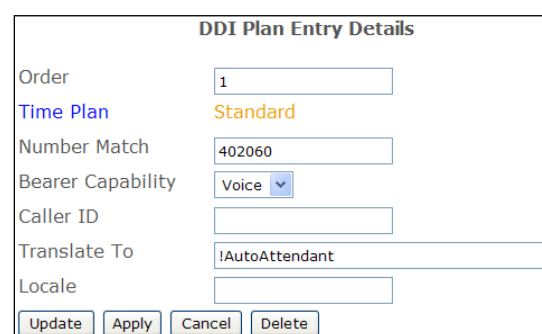|	
Time Plan	Standard
Call Server	
Number Match	777
Action	Dial
Translate To	!AutoAttendant
Translate CLI To	
LCR Plan	Standard

[Update] [Apply] [Cancel] [Delete]

Please refer to the Working with Dial Plans section from page 114 for further details on creating a Dial Plan entry.

Route external calls to the Auto Attendant

To route an incoming DDI number to the Auto Attendant a DDI Plan entry should be created with !AutoAttendant entered in the Translate To field, similar to this example:

DDI Plan Entry Details

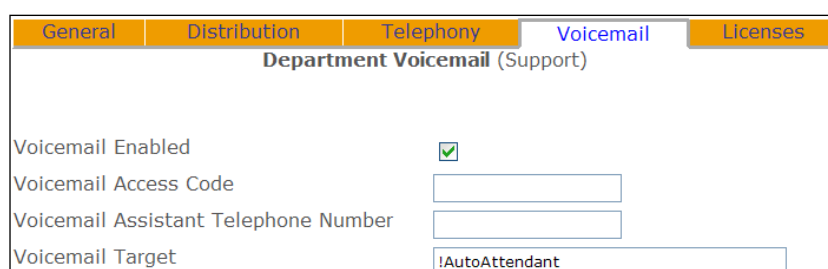|	
Order	1
Time Plan	Standard
Number Match	402060
Bearer Capability	Voice
Caller ID	
Translate To	!AutoAttendant
Locale	

[Update] [Apply] [Cancel] [Delete]

Please refer to the Working with DDI Plans section from page 165 for further details on creating a DDI Call Plan entry.

Routing a Department call to the Auto Attendant

If a call to a Department is not answered the caller can be presented with the Auto Attendant instead of being forwarded to voicemail. The Max No Answer Time before Voicemail option must be set as described on page 211.

1   In Manager select Departments
2   From the Departments list select the Department required
3   Select the Voicemail page
4   Within the Voicemail Target field enter !AutoAttendant.

General	Distribution	Telephony	Voicemail	Licenses

Department Voicemail (Support)

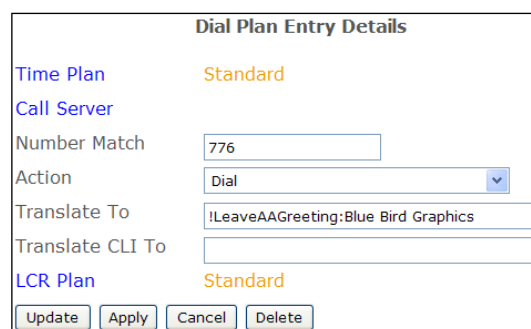|	
Voicemail Enabled	☑
Voicemail Access Code	
Voicemail Assistant Telephone Number	
Voicemail Target	!AutoAttendant

Creating an Auto Attendant for a Company

Each Company configured on the **max**imiser system can have a separate Auto Attendant as follows:

1. In Manager select Manage Company
2. Select the Company to manage (the Company name will appear in the title bar of Manager)
3. Select Auto Attendant
4. Create the Auto Attendant as described on page 222.

5. Record a greeting to announce the options using a Dial Plan entry as explained on page 223, however in the Translate To field enter !LeaveAAGreeting:<company name>, similar to this example.

Dial Plan Entry Details

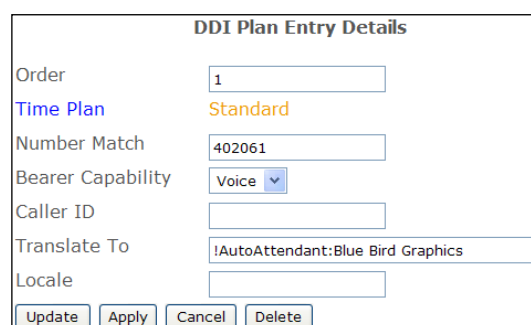|	
Time Plan	Standard
Call Server	
Number Match	776
Action	Dial
Translate To	!LeaveAAGreeting:Blue Bird Graphics
Translate CLI To	
LCR Plan	Standard

Update   Apply   Cancel   Delete

6. Create a DDI Plan entry as explained on page 224, however in the Translate To field enter !AutoAttendant: <company name>, similar to this example.

DDI Plan Entry Details

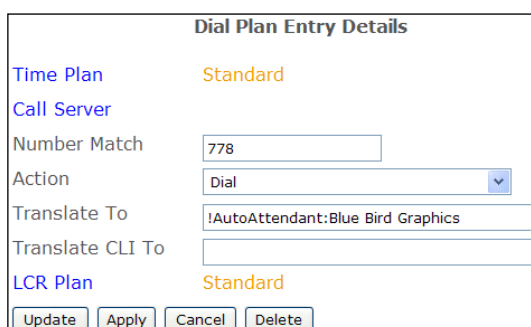|	
Order	1
Time Plan	Standard
Number Match	402061
Bearer Capability	Voice
Caller ID	
Translate To	!AutoAttendant:Blue Bird Graphics
Locale	

Update   Apply   Cancel   Delete

7. To test the Auto Attendant internally create a Dial Plan entry as explained on page 224, however in the Translate To field enter !AutoAttendant: <company name>, similar to this example.

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	778
Action	Dial
Translate To	!AutoAttendant:Blue Bird Graphics
Translate CLI To	
LCR Plan	Standard

Update   Apply   Cancel   Delete

For further information please refer to the Working with Companies section from page 155 for further details.
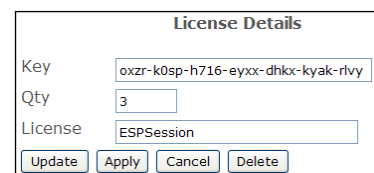
Creating VXML Scripts

Multi-level auto-attendants can be built using the VXML Scripts section in Manager incorporating features such as availability based routing, dial by extension, out of hours routing and text-to-speech. VXML is an open standard, once generated scripts can be edited manually to include custom features such as database interaction.

The configuration and set up of a VXML Script is stored on the internal Web Server. For information on accessing the Web Server please refer to page 184.

Licensing

An ESPSession licence is required to use this feature. One licence per channel required should be purchased and entered on the database.
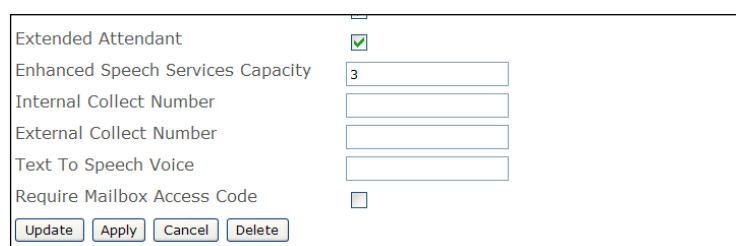
Please refer to the Installing Licences section from page 35 for further information.

The number of ESP channels to be used should be entered in the relevant Voicemail Port within the Enhanced Speech Services Capacity field and the Extended Attendant field should be ticked as shown in the following diagram.

Please refer to the Voicemail Ports section from page 188 for further information.
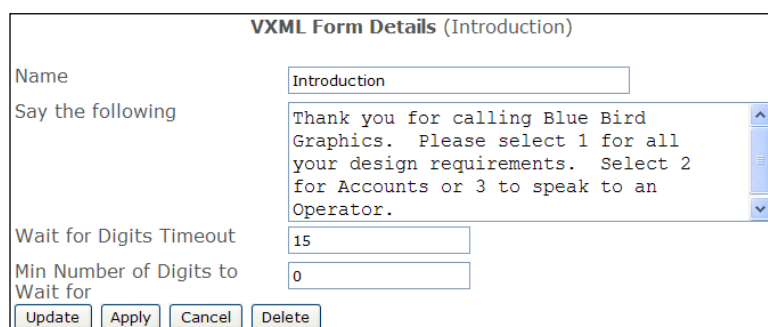
Creating a simple Auto Attendant

Firstly, it is necessary create a VXML Unit, this is the file that will contain the complete configuration of the auto attendant.

1. In Manager select VXML Scripts
2. Within the VXML Scripts list select the Add button
3. In the Name field enter a name to describe this script using alpha-numeric characters only, do not use any punctuation or spaces and begin with a letter. (This will be the name of the .php file that will be created on the internal Web Server.)
4. Select Apply

The next step is to create the Forms that will be used by the VXML Script. A VXML Script can contain multiple Forms and each Form contains the menu of choices that will be presented to the caller.

5. Under the Forms section select the Add VXML Form button
6. In the Name field enter a name to describe this Form using alpha-numeric characters only, do not use any punctuation or spaces and begin with a letter.
7. In the Say The Following field enter either the text or .wav file to be played to the caller. If text is used this will be converted to speech. If a .wav file is used this must be stored on the internal Web Server and formatted as CCITT A-Law Mono 8 bit or 16 bit.
8. If the caller does not make a selection by default the above message will be repeated after 10 seconds. However if you wish to increase this time, in the Wait for Digits Timeout field enter the number of seconds this Form should wait for the caller to make a selection.

9   In the Min Number of Digits to Wait for field enter the number digits this form should wait for before executing an action, if required

10   Select Apply.

VXML Form Details (Introduction)

Name	Introduction
Say the following	Thank you for calling Blue Bird Graphics. Please select 1 for all your design requirements. Select 2 for Accounts or 3 to speak to an Operator.
Wait for Digits Timeout	15
Min Number of Digits to Wait for	0

Update | Apply | Cancel | Delete

Within each Form the menu of options available to a caller is created as follows:

11   Within the Menu Items section select the Add VXML Menu Item button

12   The Number Match field should contain the digit to be dialled by the caller to make the required selection. This field can contain numbers 0-9, * and #. Therefore, use the default entry or edit as required.

13   In the Say The Following field enter either the text or .wav file to be played to the caller. If text is used this will be converted to speech. If a .wav file is used this must be stored on the internal Web Server and formatted as CCITT A-Law Mono 8 bit or 16 bit.

14   In the Delay Time After Speaking field enter the number of seconds that this item should wait before transferring the caller, if required.

15   In the Transfer To field enter the extension number to which the caller will be transferred.

16   Select Update

17   Repeat steps 11 to 16 to create all the menu items required.

Menu Items

Add VXML Menu Item

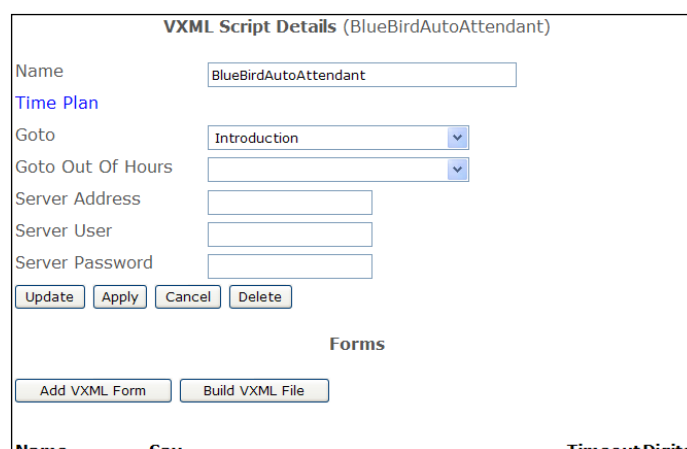| Number	Say	Delay	Transfer	Timeout	Goto	OnBusy	OnNoAnswer
1	Please hold your call is being transferred to our Design team.	0	7001	0			
2	Please hold your call is being transferred to our Accounts department.	0	7005	0			
3	Please hold while we connect you to an Operator	0	8000	0			

18   Select Update when ready. You will be returned to the VXML Script Details page.

Once all the required Forms and Menu Items have been created the first Form to be presented to a caller can be configured:

19   Within the VXML Script Details page from the Go To list box select the Form to be used when the caller first accesses this script.
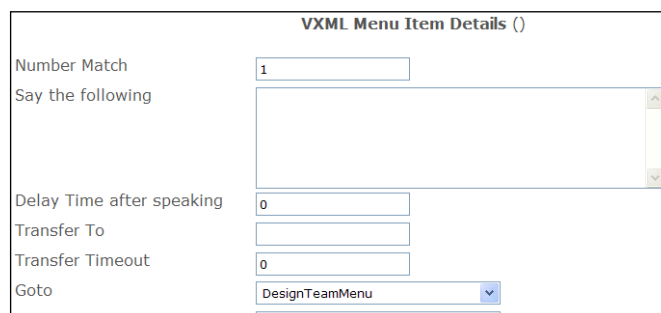
20   Select Apply

Finally, it is necessary to build the VXML file which will create a .php file on the internal Web Server with the same name as the VXML Script. This step must be taken each time changes are made to the script.

21   Within the VXML Script Details page select the Build VXML File button.

Creating a multi-layer Auto Attendant

Instead of transferring the caller to an extension a Menu Item can be configured to present the caller with another menu of options. The Form containing the required menu should be created first as described above and then selected from the Go To list box of the required Menu Item as shown in the following diagram.

Please note that an entry in the Transfer To field takes priority over the Go To field.

Using a default Menu Item

If a caller does not make a selection from the options presented a Menu Item can be used to automatically transfer the caller to a specified extension or present another Form. This can be configured as follows:

1   Within the required VXML Script Details page select the Form to be configured.
2   In the Wait for Digits Timeout field enter the number of seconds this Form should wait for the caller to make a selection. By default this is set to 10 seconds.
3   Select Apply
4   Select the Add VXML Menu Item button
5   In the Number Match field delete any entry in this field and leave the field blank. This will be displayed as Any in the VXML Form Details page.
6   If the caller is to be transferred to an extension number enter this in the Transfer To field or if the caller is to be presented with a Form select this from the Go To list box.

7   Select Update or Apply when ready.

Using a Wild Card in a Menu Item

Where a large number range is to be used within a Form, for example when offering a caller the option to dial an extension number if known, a question mark (?) in the Number Match field of a Menu Item will match any number entered, as shown in the example below.
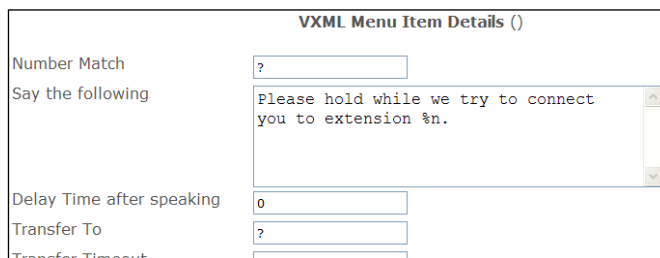
Please note that this option will allow callers to dial any number. It is therefore recommended that a Dial Plan is added to the relevant Voicemail Port to restrict the type of numbers that can be dialled, eg block international calls, restrict to internal numbers only, etc.

Using Variables

The following variables can be used in the "Say" fields to provide the following information:

%n = number (eg the caller's selection or Transfer To extension no)
%s = source number of the caller
%c = caller's name if the caller's CLI matches an entry within the Contacts database
%o = originator's caller ID

as per this example:
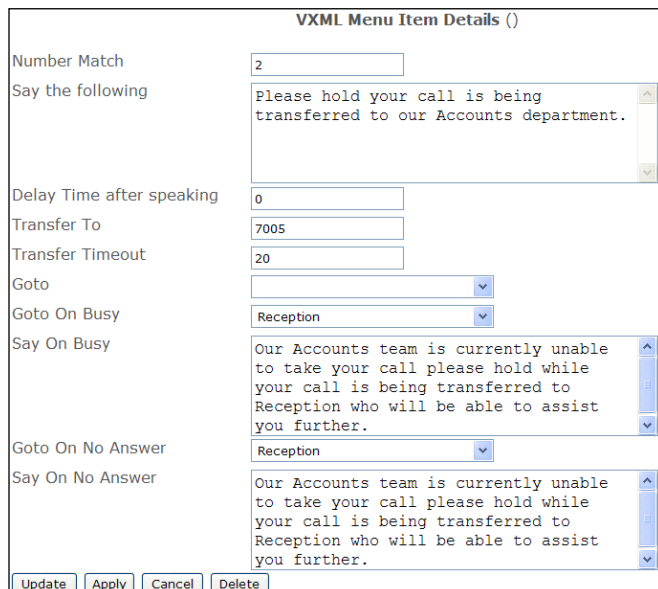
Routing a Caller on Busy or No Answer

When the Transfer To extension number is busy and/or not answered the caller can be presented with an alternative Form giving, for example, another menu of options or automatically transferring the caller to another extension. The Form(s) containing the required menu options should be created first and then the relevant Menu Item should be configured as follows:

1   Open the required Menu Item
2   In the Transfer Timeout field enter the number of seconds required before the Transfer To extension is considered not answered.
3   From the Go To On No Answer list box select the Form to be presented to the caller after the Transfer Timeout.
4   In the Say On No Answer field enter the text or .wav file to be played to the caller before presenting the Go To On No Answer Form.
5   From the Go To On Busy list box select the Form to be presented to the caller when the Transfer To extension number is busy.
6   In the Say On Busy field enter the text or .wav file to be played to the caller before presenting the Go To On Busy form.
7   Select Update or Apply when ready

Using an Out of Hours Form

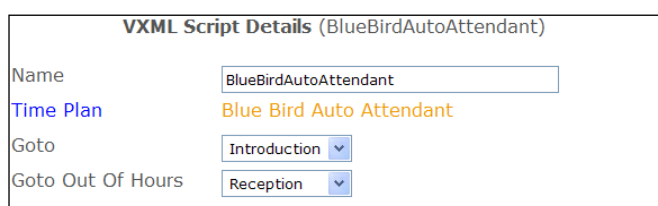A Time Plan can be used to determine the day and time when the Form displayed in the Go To field will be used. Outside of these hours a caller can presented with an alternative Form as specified in the Go To Out of Hours field. The Form and Time Plan should be created first (for further information on creating a Time Plan please refer to page 148) then the required VXML Script should be configured as follows:

1 From the VXML Script List select the required VXML Script.
2 Select the Time Plan field
3 From the Select Time Plan list select the required Time Plan.
4 From the Go To Out of Hours list box select the required Form to be presented to a caller when the call is made outside the hours specified by the Time Plan.
5 Select Update or Apply when ready.

VXML Script Details (BlueBirdAutoAttendant)

Name	BlueBirdAutoAttendant
Time Plan	Blue Bird Auto Attendant
Goto	Introduction
Goto Out Of Hours	Reception

Using the Auto Attendant Internally
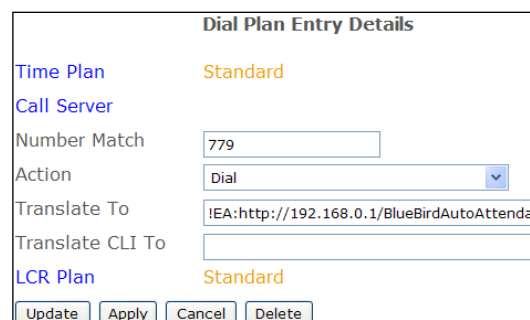
To test or use the Auto Attendant internally a Dial Plan entry can be created similar to this example:

The Translate To field should contain –

!EA:http://<ip address of the call server>/<unitname>.php

eg !EA:http://192.168.0.1/BlueBirdAutoAttendant.php

or !EA:http://127.0.0.1/BlueBirdAutoAttendant.php
to use the local Call Server.

Dial Plan Entry Details

Time Plan	Standard
Call Server	
Number Match	779
Action	Dial
Translate To	!EA:http://192.168.0.1/BlueBirdAutoAttenda
Translate CLI To	
LCR Plan	Standard

Route an external call to a VXML Script
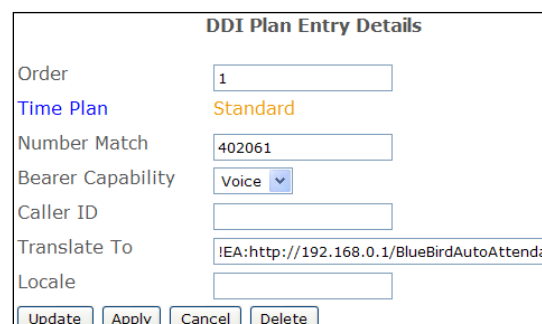
To route an incoming DDI number to a VXML Script a DDI Call Plan entry should be created similar to this example.

The Translate To field should contain –

!EA:http://<ip address of the call server>/<unitname>.php

eg !EA:http://192.168.0.1/BlueBirdAutoAttendant.php

or !EA:http://127.0.0.1/BlueBirdAutoAttendant.php
to use the local Call Server.

DDI Plan Entry Details

Order	1
Time Plan	Standard
Number Match	402061
Bearer Capability	Voice
Caller ID	
Translate To	!EA:http://192.168.0.1/BlueBirdAutoAttenda
Locale	

Call Recording

The voicemail application provides the ability to automatically record calls on the **max**imiser system. This applies to incoming calls to Users, Departments, specific DDI numbers and outgoing calls made by Users. In addition, Users can manually record any call they wish via their PCS 580/570/560, PCS 410/400, PCS 100, PCS 60, PCS 50 or PCS Operators Console. The resulting recording will be normally stored as a message in their voicemail box and can be accessed in the same way as normal voicemail messages. Please refer to the relevant PCS User manual for further details. Alternatively, Call Recordings can be utilised by external applications, such as Vision or SpliceRecord.

Please note that to use the Call Recording facility:
- The relevant User or Department will require a MessageBox licence (however voicemail does not need to be enabled.)
- The Recording Server option must be enabled within the relevant Voicemail Port. Please refer to the Voicemail Ports section from page 188 for further information.
- A User's User Class field can be used to disable manual call recording. Please refer to the Configuring a User section from page 81 for further details.

Recording Departmental Calls

Each Department has the facility to record calls either for the Department as a whole or for a particular group of Users. The calls are recorded as voicemail messages for that Department and are accessed in the same way as normal voicemail messages.

To enable recording for a Department follow these steps:

1   In Manager select Departments
2   From the Departments list select the Department required
3   Select the Voicemail page
4   From the Record Mode list box select one of the following modes:
    a   Resource – at present will also record all calls
    b   Mandatory – to record all calls
    c   Random – to record a random number of calls in line with the Record Rate
    d   Trunk – to record all calls but the User's Record icon will not be activated and therefore the User will not be able to cancel the recording.
    e   Disable – calls will not be recorded
5   If using Random record mode in the Record Rate field set a figure here for the number of calls recordings are to be made within (eg setting 5 will record 1 in 5 calls)
6   Select Update or Apply when ready.

To enable recording for a Department for a specific group of Users to define the user/users that will be subject to call recording follow these steps:

1   In Manager, firstly create a Group containing the relevant Users
2   Then select Departments
3   From the Departments list select the Department required
4   Select the Voicemail page
5   Select the Record Group field
6   From the Select Group list select the Group required. This Group will appear in the Record Group field.
7   Select Update or Apply when ready.

To send inbound call recordings to an email account the following configuration can be used. (Please note that to use this facility Voicemail Email must set up first as detailed on page 207.)

1   In Manager select Departments
2   From the Departments list select the Department required
3   Select the Voicemail page
4   Select the Add Voicemail Contact
5   In the Telephone Number field enter the email address where the recordings are to be sent.
6   From the Type list box select ArchiveEmail
7   From the Method list box select Deliver and Delete or Deliver
8   Select Update or Apply when ready.

If the Deliver and Delete Method is used the recording will be immediately deleted from the voicemail server. If the Deliver Method is used the recording will be marked as old on the voicemail server and deleted after 7 days.

Recording Inbound Calls to a User

Automatic recording of Inbound Calls can be enabled on a User by User basis. The calls are recorded as a voicemail message for that User and are accessed in the same way as normal voicemail messages.

To enable recording for a User follow these steps:

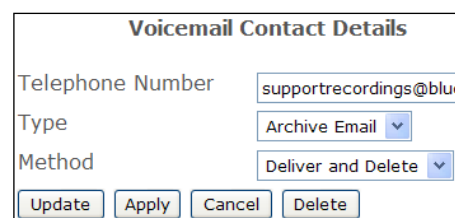1   In Manager select Users
2   From the Users list select the User required
3   Select the Voicemail page
4   From the Record Mode list box select one of the following:
    a   Resource – at present will also record all calls
    b   Mandatory – to record all calls
    c   Random – For Future Use
    d   Trunk – to record all calls but the User's Record icon will not be activated and therefore the User will not be able to cancel the recording.
    e   Disable – calls will not be recorded
5   Select Update or Apply when ready.

To send inbound call recordings to an email account the following configuration can be used. (Please note that to use this facility Voicemail Email must set up first as detailed on page 206.)

1   In Manager select Users
2   From the Users list select the User required
3   Select the Voicemail page
4   Select the Add Voicemail Contact button
5   In the Telephone Number field enter the email address where the recordings are to be sent.
6   From the Type list box select ArchiveEmail
7   From the Method list box select Deliver and Delete or Deliver
8   Select Update or Apply when ready.

If the Deliver and Delete Method is used the recording will be immediately deleted from the voicemail server. If the Deliver Method is used the recording will be marked as old on the voicemail server and deleted after 7 days.

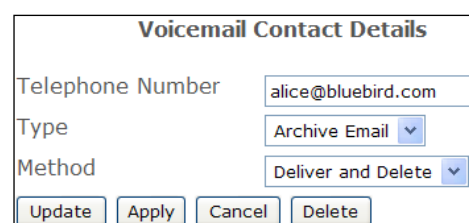## Recording Inbound Calls to a specific DDI number

Calls received via a DDI number can be automatically recorded and the resulting recordings will be saved as messages in the voicemail box called InboundRecordings. These messages can be accessed by creating a Department called InboundRecordings as follows and retrieved as described in the Voicemail for a Department section from page 209.

1. Create a Department called InboundRecordings
2. In the configuration form for this Department enter an unique extension number and assign a Message Box licence.
3. Within the Voicemail page, from the Record Mode list box select one of the following:
   a. Resource – at present will also record all calls
   b. Mandatory – to record all calls
   c. Random – for Future Use
   d. Trunk – to record all calls but the User's Record icon will not be activated and therefore the User will not be able to cancel the recording.
   e. Disable – calls will not be recorded

(For further information on creating a Department please refer to the Routing calls via a Department from page 128.)

4. To set up automatic call recording on a specific DDI number, in the relevant DDI Plan entry add an equal sign (=) before the entry in the Translate To field, similar to this example.

(For further information on creating a DDI Plan entry please refer to the Working with DDI Plans section from page 169.)

Please note that inbound recording settings for a User or Department take priority over recording settings for a DDI number.

Inbound Call Recording for a Company

Where the Companies feature is being used on the system and the inbound call recordings on different DDI numbers are to be stored in separate voicemail boxes each Company can be configured with a specific Inbound Record Mailbox.

(For further information on using the Companies feature please refer to page 155.)

In the Company Details configuration form enter the name of the User or Department to be used to store incoming call recordings in the Inbound Record Mailbox field, similar to the following example. This User or Department must be configured for call recording as per the InboundRecordings department described above. If this field is empty the resulting message will be stored in "InboundRecordings" as explained above.

In the relevant DDI Plan entry add an equal sign (=) followed by the name of the Company in square brackets before the entry in the Translate To field, similar to this example.

All calls received on this DDI number will be recorded into the voicemail box specified in the Inbound Record Mailbox for the Company specified in the square brackets.

Recording Outbound User Calls

All outbound external calls made from any handset on the system can be automatically recorded. The resulting messages are stored in a voicemail box called "OutboundRecordings". These messages can be accessed by creating a Department of the same name, as follows. To access these messages please refer to the Voicemail for a Department section from page 209.

> 1   Create a Department called OutboundRecordings
> 2   In the configuration form for this Department enter an unique extension number and assign a Message Box licence.
> 3   Within the Voicemail page, from the Record Mode list box select one of the following:
>   a   Resource – at present will also record all calls
>   b   Mandatory – to record all calls
>   c   Random – for Future Use
>   d   Trunk – to record all calls but the User's Record icon will not be activated and therefore the User will not be able to cancel the recording.
>   e   Disable – calls will not be recorded

(For further information on creating a Department please refer to the Routing calls via a Department from page 128.)

> 4   To set up automatic outbound call recording create a Dial Plan entry in the relevant Dial Plan similar to this example.

(For further information on creating a Dial Plan entry please refer to the Working with Dial Plans section from page 114.)

If the User making an outbound external call via the above Dial Plan is a member of a Company the resulting message will be stored in the voicemail box specified in the Outbound Record Mailbox field in the relevant Company's configuration form. This can be either a Department's or User's voicemail box. This User or Department must be configured for call recording as per the OutboundRecordings department described above. If this field is empty the resulting message will be stored in "OutboundRecordings" as explained above.

Company Details	
Name	Blue Bird Graphics
Description	Top floor
Park Timeout	0
Outbound Record Mailbox	Blue Bird
Inbound Record Mailbox	
Update Apply Cancel Delete	
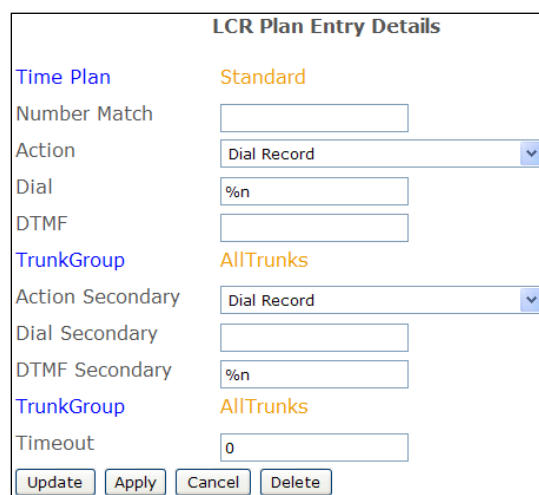
Recording all outbound calls

To automatically record ALL outbound external calls change the Action field of the default entry within the relevant LCR Plan from Dial to DialRecord. The resulting messages are stored in a voicemail box called "OutboundRecordings" which can be accessed by creating a Department of the same name as described above.

To access these messages please refer to the Voicemail for a Department section from page 209.

LCR Plan Entry Details	
Time Plan	Standard
Number Match	
Action	Dial Record
Dial	%n
DTMF	
TrunkGroup	AllTrunks
Action Secondary	Dial Record
Dial Secondary	
DTMF Secondary	%n
TrunkGroup	AllTrunks
Timeout	0
Update Apply Cancel Delete	
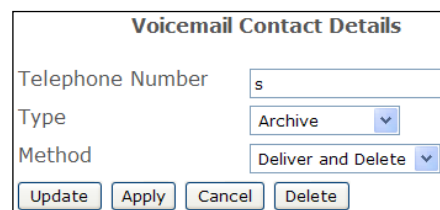
Using Call Recording with SpliceRecord

Each User and/or Department can be configured to send the recordings to SpliceRecord as follows:

1   In Manager select Users or Departments
2   From the list select the User or Department required
3   Select the Voicemail page
4   Select the Add Voicemail Contact button
5   Add some text to the Telephone Number field – this is not acted upon but there must be some form of entry in the field.
6   From the Type list box select Archive
7   From the Method list box select Deliver and Delete or Deliver
8   Select Update or Apply when ready.

Voicemail Contact Details	
Telephone Number	s
Type	Archive
Method	Deliver and Delete
Update Apply Cancel Delete	

If the Deliver and Delete Method is used the recording will be immediately deleted from the voicemail server. If the Deliver Method is used the recording will be marked as old on the voicemail server and deleted after 7 days.

Using Call Recording with Vision

For information on how to set up Call Recording with Vision please refer to the Vision Installation and Configuration Manual.

Maintaining and Troubleshooting Voicemail

Each User's and Department's voicemail box can be viewed and maintained by accessing the Mailbox directory on the Call Server as follows:

Via your browser:

1   Enter ftp://voice:<maintainerpassword>@<ip address>,   eg ftp://voice:password@192.168.0.1
2   Each voicemail box is displayed

For information on how to set the Maintainer password used to access the Mailbox directory via FTP please refer to the System Passwords section from page 251.

Via the Telnet:

1   Telnet on to the system using Root access as described in the Root Access to a Call Server section on page 252
2   Enter cd /voicemail/Mailbox, then press Enter
3   Enter ls to view each mailbox

Identifying a User's and Department's Voicemail box

Each User is identified on the database by a GUID (Global Unique Identifier). This GUID is used to identify the User's voicemail folder in the Mailbox directory.

A list of each User's, Department's and Company's GUIDs are displayed within Manager by selecting Utilities and then Voicemail GUIDs.

eb3c0700-3386-e8
13-8567-0007d900
1823

User count = 8 Department count = 3 Company count = 1 - 12 total		
3fca0700-3386-e813-8567-0007d9001823	Alice Barker	2006
4ae50700-3386-e813-8567-0007d9001823	Emma Marks	2007
eb3c0700-3386-e813-8567-0007d9001823	Jane Brown	2001
47140600-2d86-e813-8567-0007d9001823	Main	8000
9daf0700-3386-e813-8567-0007d9001823	Michael Smith	2005
f2af0400-2d86-e813-8567-0007d9001823	MyCompany	
b0930700-3386-e813-8567-0007d9001823	Peter Upton	2004
96990000-88ba-e813-8567-0007d9001823	Reception	8002
f5010800-3386-e813-8567-0007d9001823	Rose Barrett	2008
d1ec0400-acb9-e813-8567-0007d9001823	Sales Calls	8001
b3780700-3386-e813-8567-0007d9001823	Sam Turner	2003
115b0700-3386-e813-8567-0007d9001823	Simon Jones	2002

Select Back from your web browser to return to the Manager.

The User or Department's Daily Greeting is stored as Greeting.wav and the permanent greeting is stored as GreetingPermanent.wav.

View the number of messages for all Users and Departments

The status of each User's and Department's voicemail box can be viewed in Manager by selecting Utilities and then Voicemail Status. Each User and Department is listed together with the number of new messages followed by the total number of messages currently stored in their voicemail box. Each User and Department with one or more new messages waiting is displayed in white.

Select Back from your web browser to return to the Manager.

View the number of MessageBox licences assigned

A list of Users and Departments that have been assigned MessageBox licences can be viewed in Manager by selecting Utilities and then Mailbox Licences.

Select Back from your web browser to return to the Manager.

Back up voicemail

A back up of the internal voicemail server will be required to ensure that when, for example, a faulty Call Server is replaced all the Users' and Departments' messages and greetings are restored on the new Call Server. This can be done by running the following back up script.

1   Telnet on to the Call Server via Root as described in the Root Access to a Call Server section on page 252.
2   Type /ftp/backup, then press Enter
3   This will create a file called voicemail.tar0 within in the ftp directory.
4   Using FTP via a browser or Command Prompt log in as Admin on the relevant Call Server as described in the Admin Access to a Call Server section from page 252.

5   If you are using a browser the voicemail.tar0 file can be copied or moved to any location as
     required.
     or
     If you are using the Command Prompt use the "get" command to download the required file, eg
     get voicemail.tar0, and place it in the root, eg C:\. The Transmission Complete message will be
     received when the transfer is complete. Enter Quit when finished.

This back up script will also back up the internal web server, please refer to the Internal Web Server
section from page 184 for further details.

Restore voicemail

1   Make a copy of voicemail.tar0 and rename it voicemail.tar. Copy this file in to the ftp directory on
     the Call Server as follows.
2   Using FTP via a browser or Command Prompt log in as Admin on the relevant Call Server as
     described in the Admin Access to a Call Server section from page 252.
3   If you are using a browser place voicemail.tar into this area
     or
     If you are using the Command Prompt use the "put" command, eg put voicemail.tar, to copy
     voicemail.tar from the root, eg C:\, to this area. Once copied the Transmission Complete message
     will be received
4   Reboot the Call Server and all the voicemail messages and greetings will be restored.

Importing and Exporting the Configuration

Area Codes, Contacts, Dial Plans and DDI Call Plans can be imported and exported using .csv files to assist with the management of these entries or to use with other systems. Users can be exported to a .csv file to assist with the management of these records.

1   In Manager select Utilities.
2   Selecting File Import/Export

It is recommended that a backup of your configuration is made before using this feature. Please refer to page 255 for further details.

Area Codes

Import
A list of all UK Area Codes is provided with the system. This file is called areacodes.txt and is stored in the ftp directory of the Call Server. This area can be accessed via a browser or Command Prompt as described in the Admin Access to a Call Server section on page 252.

To import the entries contained in this file:
1   Select the Import button
2   Confirmation that each entry has been added to the database will be displayed.
3   Select Continue at the bottom to return to the File Import/Export screen.

When importing Area Codes if a duplicate name is found a dot will be added until the name is unique, eg if three Area Codes for Yeovil are imported, one will be called "Yeovil", the next "Yeovil." and the next "Yeovil.."

The Import function will always add entries to the database; therefore if you accidentally press the Import button twice the entries will be added twice. Use the Remove button to correct this.

If you wish to import your own Area Codes list ensure the file is saved in a .csv format and called areacodes.txt. Use FTP to copy this file into the ftp directory. You may wish to make a copy of the existing areacodes.txt before you do this.

Remove
The Remove button will delete all entries from the database that match entries found in areacodes.txt. Additional entries created manually will not be deleted. Confirmation that each entry has been deleted from the database will be displayed. Select Continue at the bottom to return to the File Import/Export screen.

Export
The Export option will copy all Area Codes entries contained in the database to a file called areacodes.out.txt. If an existing areacodes.out.txt already exists this will be overwritten. You may wish to make a copy of this file before performing an Export.

If any trailing dots are found in the Name field these will be removed.
If you wish to open areacodes.out.txt in Microsoft Excel, for example, change the file extension from ".txt" to ".csv".

You may wish to use this file as a means of changing the order the Area Code entries are displayed in the database. Make the changes required, save the file as areacodes.txt and copy it into the ftp directory, then Import the file.

Export & Delete
This option will export all Area Code entries to areacodes.out.txt and delete the entries from the database.

Contacts

Import
The Import option allows Contact entries stored on an existing database to be imported into the configuration. The field names must match the following syntax:

company, name, phone, fax, mobile, home, spare1, spare2, ambiguous, mail, firstname, lastname, description, department, job, street, city, county, postcode, country, ourAccount, theirAccount, URL, autoURL

Save the file storing this information in a .csv format and name it as contacts.txt. Use FTP via a browser or Command Prompt, as described in the Admin Access to a Call Server section on page 252, to copy this file into the ftp directory of the Call Server,

To import the entries into the configuration:
1   Select the Import button
2   Confirmation that each entry has been added to the database will be displayed.
3   Select Continue at the bottom to return to the File Import/Export screen.

If a duplicate name is found a dot will be added until the name is unique, eg if three Contacts called J Smith are imported, one will be called "J Smith", the next "J Smith." and the next "J Smith.."

Any fields that are not recognised by the system will be ignored. The order of the fields in contacts.txt is irrelevant, the system will recognise the field name. All telephone numbers will be imported in a format that can be dialled by the system, ie brackets, spaces etc are removed.

If a Contact's address consists of more lines than just Street, Town and County, fields labelled "address1", "address2", "address3" and "address4" will be imported and displayed in the Street field separated by a comma. These fields will replace any field labelled "street".

The Import function will always add entries to the database, therefore if you accidentally press the Import button twice the entries will be added twice. Use the Remove button to correct this.

Remove
The Remove button will delete all entries from the database that match entries found in contacts.txt. Additional entries created manually will not be deleted. Confirmation that each entry has been deleted from the database will be displayed. Select Continue at the bottom to return to the File Import/Export screen.

Export
The Export option will copy all Contact entries contained in the database to a file called contacts.out.txt. If an existing contacts.out.txt already exists this will be overwritten. You may wish to make a copy of this file before performing an Export.

If any trailing dots are found in the Name field these will be removed.

If you wish to open contacts.out.txt in Microsoft Excel, for example, change the file extension from ".txt" to ".csv".

You may wish to use this file as a means of changing the order the Contact entries are displayed in the database. Make the changes required, save the file as contacts.txt and copy it into the ftp directory, then Import the file.

Export & Delete
This option will export all Contact entries to contacts.out.txt and delete the entries from the database.

Import Option
If selected when importing contacts.txt the system will create an ambiguous number by removing the last two digits from the Telephone number and replacing these with questions marks.

Export Option
If this option is selected when exporting to contacts.out.txt a space will be added to telephone numbers between the first 0 and the next digit so that Microsoft Excel, for example, will not treat the entry as a number and remove the 0.

Dial Plan

Use this Plan
This field will determine the Dial Plan into which Dial Plan entries will be imported or exported from as explained below.

Import
Files containing ready-made Dial Plan entries are provided with the system, such as dialplan.txt, dialplan520.txt, etc, and are stored in the ftp directory of the Call Server. This area can be accessed via a browser or Command Prompt as described in the Admin Access to a Call Server section on page 252.

These default Dial Plan entries can be imported into the configuration. To do this the file must be called dialplan.txt.

Dial Plan entries exported from another system can also be imported into the database, this file must be called dialplan.txt and copied in to the ftp directory as described in the Admin Access to a Call Server section on page 252. You may wish to make a copy of the existing dialplan.txt before renaming the new file.

To import the entries contained in this file:
1   Firstly determine into which Dial Plan these entries will be copied
2   Enter the name of the Dial Plan in the Use This Plan field. This Dial Plan must already exist on the database.
3   Select the Import button.
4   Confirmation that each entry has been added to the database will be displayed.
5   Select Continue at the bottom to return to the File Import/Export screen.

No checks are made for duplicate entries.

The Time Plan and LCR Plan fields may not be imported (unless dialplan.txt has been created from a file exported from the same system, see below), unless you select the relevant option (see below) and must be manually configured.

The Import function will always add entries to the database; therefore if you accidentally press the Import button twice the entries will be added twice. Use the Remove button to correct this.

Remove

The Remove button will delete all entries from the Dial Plan entered in the Use This Plan field that match entries found in dialplan.txt. Additional entries created manually will not be deleted. Confirmation that each entry has been deleted from the database will be displayed. Select Continue at the bottom to return to the File Import/Export screen.

Export

The Export option will copy all Dial Plan entries contained in the Dial Plan entered in the Use This Plan field to a file called dialplan.out.txt. If an existing dialplan.out.txt already exists this will be overwritten. You may wish to make a copy of this file before performing an Export.

If you wish to open dialplan.out.txt in Microsoft Excel, for example, change the file extension from ".txt" to ".csv".

You may wish to use this file as a means of changing the order the Dial Plan entries are displayed in the Dial Plan. Make the changes required, save the file as dialplan.txt and copy it into the ftp directory. Ensure the required Dial Plan has been entered in the Use This Plan field and Import the file.

Export & Delete

This option will export all Dial Plan entries from the specified Dial Plan to dialplan.txt and delete the entries from the specified Dial Plan.

Import Option (If file came from recent export..)

If the dialplan.txt file was created from a file exported from the same system then this option can be selected and Time Plan and LCR Plan entries will be imported.

Import Option (Guess at extra fields..)

This option can be used to 'second guess' the entries for Time Plan and LCR Plan fields provided there is already one entry in the Dial Plan with these fields entered. All entries imported will then have the same as the existing entry.

Export Option

If this option is selected when exporting to dialplan.out.txt a space will be added to telephone numbers between the first 0 and the next digit so that Microsoft Excel, for example, will not treat the entry as a number and remove the 0.

DDI Plan

Use this Plan

This field will determine the DDI Plan into which DDI Plan entries will be imported or exported from as explained below.

Import

DDI Plan entries exported from the same system or from another system can be imported into a specified DDI Plan. The file containing these entries must be called ddiplan.txt and copied into the ftp directory of the Call Server. This area can be accessed via a browser or Command Prompt as described in the Admin Access to a Call Server section on page 252.

To import the entries contained in this file:

1   Firstly determine into which DDI Plan these entries will be copied
2   Enter the name of the DDI Plan in the Use This Plan field. This DDI Plan must already exist on the database.
3   Select the Import button.
4   Confirmation that each entry has been added to the database will be displayed.
5   Select Continue at the bottom to return to the File Import/Export screen.

No checks are made for duplicate entries.

The Time Plan field will not be imported unless ddiplan.txt has been created from a file exported from the same system and the Import Option is selected, see below.

The Import function will always add entries to the database; therefore if you accidentally press the Import button twice the entries will be added twice. Use the Remove button to correct this.

Remove

The Remove button will delete all entries from the DDI Plan entered in the Use This Plan field that match entries found in ddiplan.txt. Additional entries created manually will not be deleted. Confirmation that each entry has been deleted from the database will be displayed. Select Continue at the bottom to return to the File Import/Export screen.

Export

The Export option will copy all DDI Plan entries contained in the DDI Plan entered in the Use This Plan field to a file called ddiplan.out.txt. If an existing ddiplan.out.txt already exists this will be overwritten. You may wish to make a copy of this file before performing an Export.

If you wish to open ddiplan.out.txt in Microsoft Excel, for example, change the file extension from ".txt" to ".csv".

You may wish to use this file as a means of changing the order the DDI Plan entries are displayed in the DDI Plan. Make the changes required, save the file as ddiplan.txt and copy it into the ftp directory. Ensure the required DDI Plan has been entered in the Use This Plan field and Import the file.

Export & Delete

This option will export all DDI Plan entries from the specified Dial Plan to dialplan.txt and delete the entries from the specified Dial Plan.

Import Option

If the ddiplan.txt file was created from a file exported from the same system then this option can be selected and Time Plan field will be imported.

The Time Plan Name is stored as a number, which is unique to each system and therefore cannot be exported between systems.

Export Option

If this option is selected when exporting to ddiplan.out.txt a space will be added to telephone numbers between the first 0 and the next digit so that Microsoft Excel, for example, will not treat the entry as a number and remove the 0.

Users

Export

The Export option will copy all Users stored on the database to a file called users.out.txt. This file will be stored in the ftp directory of the Call Server, which can be accessed via FTP using a browser or Command Prompt as described in the Admin Access to a Call Server section on page 252.

If an existing users.out.txt already exists this will be overwritten. You may wish to make a copy of this file before performing an Export.

If you wish to open users.out.txt in Microsoft Excel, for example, change the file extension from ".txt" to ".csv".

This file can be used for managing a list of Users as a database or spreadsheet application will allow you to quickly and easily search, for example, which Users have been configured with a MessageBox licence.

Export Option

If this option is selected when exporting to users.out.txt a space will be added to telephone numbers between the first 0 and the next digit so that Microsoft Excel, for example, will not treat the entry as a number and remove the 0.

Modify Users

This feature of Manager allows the Users configuration to be exported as a .csv file and modified via Telnet or an application such as MS Excel.

Download the Users configuration to your PC

1   From Manager select Utilities
2   Select Modify Users
3   Select Dump (located in the top right hand corner)
4   Enter the range of extension numbers for the User configurations that you wish to download

5   Select the Dump button
6   The configuration will be placed in a file called modify.out.csv and stored on the Web Server in the Download folder (to access this folder please refer to the Internal Web Server section on page 184).
7   A list of the extensions selected and "Done" will appear once this task has been completed. (Modify.out.csv can now be edited locally via Telnet and "vi".)
8   To copy this file to your PC select Download
9   Specify where you wish to save this file on your PC and wait until the task is complete.
10 This file can now be edited via an application such as MS Excel.

Please note:

- MS Excel will remove leading zeros from numbers therefore it is recommended that you to format all columns containing telephone numbers as text.
- This feature does not download entries within the User's Speed Dial page.

Use the Refresh option (at the top right hand corner of Manager) to return to the first Modify Users screen.

Upload the amended Users configuration to the database

1 From Manager select Utilities
2 Select Modify Users
3 Select Upload
4 Select Browse to select the .csv file to be uploaded
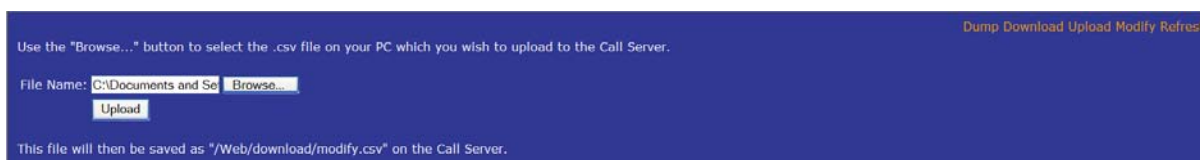
5 Select the Upload button and the file will be saved as modify.csv in the Download folder on the Web server.
6 To make the changes to the configuration on the database select Modify
7 Select Continue from the message displayed
8 Details of the upload will be displayed followed by "Done" to confirm that changes have been completed.

Please note: when certain fields are amended in this way Users' phones will re-register with the system and interrupt active calls.

Use the Refresh option (at the top right hand corner of Manager) to return to the first Modify Users screen.

Data Routing

The **max**imiser Call Server has the ability to route IP data packets either onto the Local Area Network (LAN) for onward routing to a remote site via a gateway router or out to a Wide Area Network (WAN) via ISDN dial up.

IP routes for routing onto the LAN are entered under the Module whilst IP routes for routing via a WAN connection need to be entered under the relevant WAN Link.

Use of the industrial standard PPP (Point to Point Protocol) is used to allow connectivity to third party routers supporting PPP.

Bandwidth on demand data links over ISDN allow call charges to be kept to a minimum as the link will only be active when data is flowing. Several ISDN calls can be linked together to increase bandwidth of the connection. These links also use the industrial standard PPP/MLPPP.

Please note:
- At present, due to the bandwidth requirements for a voice call being greater than 64kbps, it is not recommended to use ISDN calls to link sites for voice traffic.
- The system does not support NAT/Firewall therefore it is recommended that you only connect 'friendly' sites and **do not** connect the **max**imiser directly to the Internet

Creating a WAN Link

1 In Manager select WAN Links
2 Select the relevant module containing the WAN port required, eg CallServer
3 Either edit the default link called WANLink and proceed from step 6,
   or
4 Select the Add button
5 Enter a Name using alphanumeric characters only, maximum of eight, eg the location of Call Server, and a description if required
6 In the Account Name Out field enter a name that will be used for access authentication by the distant end using alphanumeric characters starting with a letter
7 In the Account Password Out field enter a password that will be used for access authentication by the distant end
8 In the Account Name In field enter a name that the distant end will send for access authentication
9 In the Account Password In enter a password that the distant end will send for access authentication
10 In the Telephone Number field enter the number to dial if this is a dial up link.
11 In the DDI Telephone Number field enter the DDI number that is to be used for dial in access to this Call Server or enter * for any data call or if left blank will deny ISDN access into this Call Server
12 Select Apply
13 Select the Time Plan field and from the Select Time Plan list select the time plan required. This will be displayed in the Time Plan field.
14 Select Apply
15 Select the Add Trunk button
16 Select the Trunk field
17 Select the module containing the Trunk to be used
18 From the Select Trunk list select the trunk required.
19 Select Update or Apply when ready.

Setting up IP Routes

At present the **max**imiser only supports static IP addresses therefore you will need to enter details of the remote network.

1 In Manager, select WAN Links
2 Select the relevant module containing the WAN Port required
3 Select the relevant link from the WANLink List
4 Select the IPRoutes page
5 Select the Add IP Route button
6 Enter the IP Address, IP Mask and Metric in the relevant fields
7 Select Update
8 Add further routes if required

IP Address Details

IP Address	192.168.107.13
IP Mask	255.255.255.255
IP Gateway	
Metric	4

Update | Apply | Cancel | Delete

Setting up PPP

Point to Point Protocol (PPP) is the industry standard used by the **max**imiser to allow connection to another **max**imiser or a third party router supporting the same industry standard.

1 In Manager select WAN Links
2 Select the relevant module containing the WAN Port required
3 Select the relevant link from the WANLink List
4 Select the PPP page
5 PPP Multilink is used for enabling or disabling the negotiation of Multilink. This would typically be enabled but some products may object to negotiating Multilink.
6 Maximum Reconstructed Reception Unit (MRRU) specifies the maximum number of octets in the Information fields of reassembled packets, typically set to 1500
7 Maximum Receive Unit (MRU) the negotiated packet size that can be received from a Multi Link PPP connection, typically set to 1500
8 Authentication Protocol is the level of security checking between the two ends.
  a   PAP – Password Authentication Protocol, provides a simple method for the peer to establish its identity using a 2-way handshake. This is done only upon initial link establishment.
  b   CHAP - Challenge-Handshake Authentication Protocol is used to periodically verify the identity of the peer using a 3-way handshake. This is done upon initial link establishment, and MAY be repeated anytime after the link has been established.
9 Authentication Method, select one of the following:
  a   Incoming - authentication is only verified in one direction
  b   Outgoing - authentication is only verified in one direction
  c   Bothway – default, recommended; authentication is verified in both directions

Setting up Bandwidth on Demand (BOD)

1 In Manager select WAN Links
2 Select the relevant module containing the WAN Port required
3 Select the relevant link from the WANLink List
4 Select the On Demand page
5 Min. Call Time specifies that the call should be up for at least the specified amount of time
6 No Session Idle Period specifies the time period a link needs to be idle before the Call Server will close the connection
7 Min. Channels Required specifies the number of ISDN Channels the Call Server should use on initial dial up.
8 Max. Channels Allowed specifies the total number of ISDN Channels the Call Server will use for this link.
9 Extra Bandwidth Threshold specifies the amount of throughput on the existing link before an extra ISDN channel is added.
10 Reduce Bandwidth Threshold specifies the level of throughput on the existing link before an ISDN channel is removed.

Maintenance & Troubleshooting

Utilities

By selecting Utilities in Manager the following facilities are available to assist with the management of the system.

Backup now
Creates a backup of the configuration stored on the database. The backup file created will be stored in the ftp directory on the Call Server. Please refer to page 255 for further details.

Ask Master
This facility is used when a Call Server is added to an existing system. Please refer to the Connecting Multiple Call Servers section from page 72 for further details.

Set Time
Use this facility to set the time used by the system. Please refer to page 30 for further details.

Call Status
This facility allows all calls currently active on the system to be viewed. Please refer to page 27 for further details.

Voicemail Status
Allows the status of each User's and Department's voicemail box to be viewed. Please refer to page 238 for further details.

File Import Export
This facility is used to import or export specific sections of the database for management in third party applications. Please refer to page 240 for further details.

Modify Users
This feature allows the Users section of the database to be exported as a .csv file and modified via a third party application. Please refer to page 245 for further details.

Warnings
A list of each reboot of the system and licence error warnings are displayed. Select Home from your web browser to return to Manager

IP Addresses
A list of the IP addresses used by each module or IP phone is displayed. Please refer to the configuration section for each module or PCS IP phone for further details.

Mailbox Licences
A list of which Users and Departments have been assigned a MessageBox licence is displayed. Please refer to page 189 for further details.

Sorted Lists
It is essential that duplicate User and Department Names and Extension Numbers are not used on a system and this feature will highlight entries where an identical name or extension number exists. It is

also useful for quickly viewing which physical analogue port or PCS a User is assigned to and ensuring that the port or PCS has not been assigned to more than one User. Please refer to the relevant sections in this manual for further information.

User Status
Where multiple Call Servers are being run on one system the User Status facility will verify to which Call Server Users are registered. For further information on working with multiple Call Servers please refer to page 72.

Voicemail GUIDs
Each User, Department and Company is identified on the database by a GUID (Global Unique Identifier) and this facility will display a list of these GUIDs. These can then be used to identify the relevant voicemail folder on the voicemail server. Please refer to page 237 for further details.

PHP Utility Pages

Various utility pages to help with the maintenance of the system, such as licence status, time plan status, move Users' location etc, are available by opening:

http://<ip address of call server>/manager/custom0.php

eg http://192.168.0.1/manager/custom0.php

System Access

When maintaining and troubleshooting the maximiser system you may need Root, Admin, Voice or Web access to the Call Server.

System Passwords

This access to the system is password protected and the default password for each is *password*. It is highly recommended that these passwords are changed as follows to ensure the security of the system.

Changing the Password for Root access

1   In Manager, select System
2   In the Diagnostic Engineering Password field enter the password required
3   Select Update or Apply when ready.

Root access is required when Telnet is used to access the root directory on the call server as explained on page 252.

Change the Password for Admin and Voice access

1   In Manager, select System
2   In the Maintainer Password field enter the password required
3   Select Update or Apply when ready.

Admin access is required when using FTP to access the ftp directory on the call server as explained on page 252.

The Voice access is required for FTP access to the Mailbox directory as described in the Troubleshooting Voicemail section from page 237.

Changing the Password for Web access

1　In Manager, select System
2　In the Web Password field enter the password required
3　Select Update or Apply when ready.

The Web access is required for FTP access to the Web directory as described in the Internal Web Server section from page 184.

Root Access to a Call Server

Root access to a Call Server will be required when using Telnet to access the root directory on the Call Server. This will assist with troubleshooting the system as described further in this section.

Via the Command prompt on a PC connected to the system

1　Enter

　telnet <ip address of Call Server > eg telnet 192.168.0.1

2　At the Login prompt enter *root*, and then press Enter
3　At the Password prompt enter the Diagnostic Engineering Password, eg password, and then press Enter
4　A hash (#) will appear indicating that the previous entries where accepted and a command can be entered. You are now at the directory /root.

For further information on the commands that can be used here please refer to the Using Telnet section from page 270.

Admin Access to a Call Server

Admin access to a Call Server will be required when using FTP to access the ftp directory on the Call Server. This will be required, for example, when setting up the music on hold feature as described in the Configuring Music Channels section from page 177, to assist with troubleshooting as described further in this section, and when importing and exporting the configuration as described the Importing and Exporting the Configuration section from page 240.

Via your browser enter:

　ftp://admin:<maintainer password>@<ip address of Call Server>

　eg ftp://admin:password@192.168.0.1

or

Via the Command prompt on a PC connected to the system

1　Enter

ftp <ip address of Call Server>   eg ftp 192.168.0.1

2   At the User prompt enter *admin*, then press Enter

3   At the Password prompt enter the Maintainer Password, eg password, then press Enter

The following FTP commands can then be used.

get a   - get file from the current directory and place in root, eg C:\
put a   - put file from root, eg C:\, into the current directory
mget B*   - will attempt to get all files from the FTP and ask you if you want them (y/n)
mode image   - switch to binary file transfer
hash   - show progress
quit   - exit
tftp   - trivial file transfer protocol eg tftp –g –l frog –r frog 192.168.0.1

Reset Switch

The Reset Switch on the front of a module can be used to assist with troubleshooting the SpliceCom system.

Orderly Shut down of the File System

It is important to shut down the file system within the Call Server before removing power.

1   Press the Reset Switch once.
2   The file system will start to shut down
3   After about 10 seconds the LEDs on the front of the Call Server will flash and turn off
4   The SpliceCom LED will flash slowly to indicate that the shut down is complete.
5   At this point the power can be removed.

An orderly shutdown of the Call Server is also required before a Reboot and when defaulting the database as explained below.

An orderly shut down of a Phone and Trunk module will also be required when performing a reboot and upgrade as explained from page 260.

Reboot (Warm Start)

A reboot of a module may be required to restore connection to the LAN, restore connection to the Call Server etc.

1   Firstly perform an orderly shut down as explained above.
2   When the SpliceCom LED is flashing press the Reset Switch once (or no more than 3 times).
3   The LED will turn off for 10 seconds and then flash rapidly as the system restarts. Alternatively wait for 60 seconds the module will automatically restart.

Defaulting the Database (Cold Start)

You may wish to erase any configuration changes that you have made to the database and return to the default configuration. Remember to make a copy of the configuration first if you wish to keep it – see page 255. This will put the configuration back to the default settings without affecting the voicemail or the web server.

1. Firstly perform an orderly shut down on the Call Server
2. Press the Reset Switch 4 times (or no more than 7 times).
3. The SpliceCom LED will flash once, flash again after 2 seconds and then turn off for 10 seconds.
4. Wait for the LED to be on solid again to indicate that the system has restarted successfully.

Upgrading the Software on a Module

The software for the Phone and Trunk Module is stored on the Call Server. Upgrade these modules as follows: (Ensure you have a LAN connection to the Call Server). For further information please refer to page 258.

1. Firstly perform an orderly shut down on the Phone or Trunk module
2. When the SpliceCom LED is flashing press the Reset Switch 8 times (or no more than 10 times)
3. The LED will flash twice, flash twice again after 2 seconds and then turn off.
4. The activity LED on the LAN port will go solid for a few seconds as the software is downloaded.
5. Shortly after the Splicecom LED will be on and the unit will register with the system again. Each port will flash the ring LED.

The software on the Call Server is upgraded via a Factory Restart or via software patch as detailed from page 258. However if you accidentally press the Reset Switch 8 times on a Call Server no action will be performed.

Reset Switch Summary

The following table gives a summary of the functionality of the Reset Switch after an orderly shut down.

No of presses		Function	LED Indicator
Min	Max		
1	3	Reboot	LED will go off, wait 10 secs, then flash rapidly during restart
4	7	Default the database	LED will flash one, repeat after 2 secs, then go off. Wait for system to come back up, LED on solid.
8	10	Upgrade software.(Not Call Server).	LED will flash twice, repeat after 2 secs, then go off. Watch LAN activity LED for download.
11	n/a	Total Erase (**SpliceCom use only**)	NOT TO BE DONE BY ANYONE OTHER THAN SPLICEOM PERSONNEL. THIS WILL RENDER THE UNIT INOPERABLE.
12+	n/a	Abort command and perform reboot	

Powering down a Call Server

It is important to shut down the file system within the Call Server before removing power.

1   On the Call Server press the Reset Switch once.
2   The SpliceCom LED will turn off for a maximum of 10 seconds while the file system closes
3   The LED will then flash slowly to indicate that the shut down is complete.
4   At this point the power can be removed.

Please note that the SpliceCom LED will flash for 60 seconds and then the Call Server will automatically reboot.

The above procedure can be performed on a Phone and Trunk Module before power down, however as neither module contains a hard disk this procedure is not essential.

Rebooting a module

A reboot may be required on a module in order to restore connection to the LAN, restore connection to the Call Server etc.

1   On the relevant module press the Reset Switch once.
2   The SpliceCom LED will turn off for a maximum of 10 seconds
3   The LED will then flash slowly to indicate that the shut down is complete.
4   Press the Reset Switch once (or no more than 3 times).
     or
     wait for 60 seconds the module will automatically restart.
5   The LED will turn off for 10 seconds and then become solid one the unit is back up.

Backing up the Database

A regular back up of the configuration of the system should be made in order to be able to restore a system quickly and easily if, for example, an error is made in the configuration and you wish to return to a previous configuration or a faulty Call Server is replaced.

Please note: remember to also back up any WAV files or .csv files stored in the ftp directory, and to back up the internal voicemail server (as described in the Troubleshooting Voicemail section from page 237) and web pages (as described in the Internal Web Server section from page 184).

Creating a back-up of the configuration

1   In Manager, select Utilities
2   Select Backup now
3   A backup of the current configuration of the database will be made.

Please note:
• A back up is also created each time the Call Server is rebooted.
• A copy of the backup file must be made before a Factory Restart, as this file will be deleted.

Access a backup file

Back up files are stored on the Call Server and can be access via FTP as follows:

1   Using FTP via a browser or Command Prompt log in as Admin on the relevant Call Server as described in the Admin Access to a Call Server section from page 252.
2   If you are using a browser the backup text file created is displayed giving the date and time that the backup was made. This can be copied or moved to any location as required.

Backup 21.08.07
143535.txt

or
If you are using the Command Prompt use the "get" command to download the required file, eg get Backup 15.01.03 092508.txt, and place it in the root, eg C:\. The Transmission Complete message will be received when the transfer is complete. Enter Quit when finished.

Restore the system from a back up file

A back up file can be sent to a Call Server as follows:

1   Make a copy of the backup text file and rename it newdatabase.txt.
2   Using FTP via a browser or Command Prompt log in as Admin on the relevant Call Server as described in the Admin Access to a Call Server section from page 252.
3   If you are using a browser place newdatabase.txt into this area
    or
    If you are using the Command Prompt use the "put" command, eg put newdatabase.txt, to copy newdatabase.txt from the root, eg C:\, to this area. Once copied the Transmission Complete message will be received
4   Perform an orderly shut down of the Call Server by pressing the Reset Switch **once**. The SpliceCom LED will slowly flash when the shut down is complete
5   Press the Reset Switch once to restart the Call Server and the newdatabase.txt file will be copied into the database.

If the file has been accepted and the database updated the newdatabase.txt file will be automatically deleted from the ftp directory.

Please note: a backup of the configuration from a 5100 Call server can not be sent to a 5108 Call Server and visa versa, as well as a back up from a 4100 Call Server can not be sent to a Remote Call Server and visa versa.

Using a back up file on a different Call Server

If a back up file is to be sent to a different Call Server when, for example, taking a pre-prepared configuration to a customer's site the following steps can be performed:

1   Make a copy of the backup file and rename it newdatabase.txt.
2   Place newdatabase.txt onto the Call Server via FTP as described in steps 2 and 3 in the above section.
3   Telnet on to the system via Root as described in the Root Access to a Call Server section on page 252
4   Enter the command /ftp/ChangeMac (You can actually enter /ftp/Ch<TAB>).

5   This will run a short script that will change the MAC address within the file newdatabase.txt with that of the new Call Server.

6   Perform a normal reboot of the Call Server either via the Reset switch or enter the command poweroff in your telnet session.

Swapping faulty units

If a module or IP phone connected to the system is faulty and is to be replaced the following steps will need to be taken. It is recommended that a like for like swap is always made, eg a 5315 Phone for a 5315 Phone, a PCS 560 with another PCS 560 and so on. Before connecting any new hardware to the system ensure that Auto Add Phones is turned off (as described in the Using Auto Add Phones section from page 34).

Call Server

If you still have access to the original Call Server make a back up of the configuration, voicemail, web pages, WAV files etc as described above. Otherwise you will need to have access to back ups created prior to the Call Server failure.

1   Replace the original Call Server module with the new one but do not connect it to the network at this time.

2   Send the back up file you wish to use to the Call Server as described in the Restore the system from a back up file section from page 256. Ensure the file is called newdatabase.txt

3   Telnet on to the system via Root as described in the Root Access to a Call Server section on page 252

4   Enter the command /ftp/ChangeMac (You can actually enter /ftp/Ch<TAB>).

5   This will run a short script that will change the MAC address within the file newdatabase.txt with that of the new Call Server.

6   Perform a normal reboot of the Call Server either via the Reset switch or enter the command poweroff in your Telnet session.

7   After the Call Server has been rebooted and its configuration has been loaded you will need to go back into the configuration (the Call Server will still be 192.168.0.1) and set the required IP address and DHCP parameters.

8   Once the Call Server is back up verify that it is running the configuration you expect before connecting it back into the network.

9   You can now restore voicemail and any web pages (as described in the Troubleshooting Voicemail section from page 237 and in the Internal Web Server section from page 184).

10   Copy any wav files required for music on hold back in to ftp directory on the Call Server (as described in the Admin access to the Call Server section on page 252.)

11   Ensure you can make calls.

Phone or Trunk Module

1   Ensure the Auto Add Phones feature is disabled within the relevant Call Server.

2   Connect the new module to the system and power up.

3   The new module will appear in the Unassigned Modules list

4   Select the module and copy its MAC address (serial number)

5   From the Modules list open the existing module and change its MAC address (serial number) for that of the new one.

6   Update this change

7   Delete the new module from the Unassigned Modules list.

IP Phone

1 Ensure the Auto Add Phones feature is disabled within the relevant Call Server.
2 Connect the new phone to the system and power up
3 The new phone will appear in the Unassigned Phones list
4 Select the phone and copy its MAC address (serial number)
5 From the Phones list open the existing phone and change its MAC address (serial number) to that of the new one.
6 Update this change
7 Delete the new phone from the Unassigned Phones list
8 The original User will be automatically logged into the replacement phone

Upgrading the System

Software upgrades will be supplied as a file such as
cs5100.patch.3.2.879.tar.gz (for the 5100 Call Server),
cs5108.patch.3.2.879.tar.gz (for the 5108 Call Server) or
callserver.patch.3.2.879.tar.gz (for the 4100 Call Server) or
rcallserver.patch.3.2.879.tar.gz (for the 4140 Remote Call Server) and
pcs580.3.2.589.tar.gz (for the PCS 580)
pcs5xx.3.2.141.tar.gz (for the PCS 560 and 570) or
pcs410.3.2.879.tar.gz (for the PCS 410).

The numbers given as eg 3.2.879 indicate the version number of the patch. Each patch is accumulative.

The patch file must be renamed before proceeding to the next stage.
• Edit the file name to remove the version number so that the files are called, eg cs5100.patch.tar.gz or pcs410.tar.gz
• If after downloading the file the name of the file has changed to, eg cs5100.patch.3.2.879.tar.tar, rename the file to change the final .tar to .gz.

Root Upgrade

If you are upgrading from version 3.2.620 or below you need to download the following file:

cs5100.root.3.2.621.tar.gz or cs5108.root.3.2.621.tar.gz or callserver.root.3.2.621.tar.gz or rcallserver.root.3.2.621.tar.gz,

and rename as
cs5100.root.tar.gz or cs5108.root.tar.gz or callserver.root.tar.gz or rcallserver.root.tar.gz

If you are upgrading a 4100 Call Server or 4140 Remote Call Server from version 3.1 or below you will need to perform a root upgrade by downloading the following file:

callserver.root.tar.gz or rcallserver.root.tar.gz

If, however, the current software version is lower than 2.2.600 you will also need to download the following file:

adminmod.patch.tar.gz (4100) or rmtcs.patch.tar.gz (4140)

Please note: A root upgrade will reset Web pages, inittab and startVpn files back to default. Please ensure you have a back up of the files required before upgrading the system. (If you require further information on these files please contact Support.)

Send a Software Patch to the Call Server

Once downloaded the patch file must be sent to the relevant Call Server via FTP as follows:

1   Using FTP via a browser or Command Prompt log in as Admin on the Call Server (as described in the Admin Access to a Call Server on page 252.)
2   If you are using a browser copy the patch file into this area
      or
      If you are using the Command Prompt use the "put" command, eg put cs5100.patch.tar.gz, to copy the patch file from the root, eg C:\, to this area. Once copied the Transmission Complete message will be received. Enter Quit when finished.

Once the patch has been sent to the Call Server reboot the Call Server as follows:

1   Perform an orderly shut down of the Call Server by pressing the Reset Switch **once**. The SpliceCom LED will slowly flash when the shut down is complete. (If you are performing a root upgrade a second reboot may be required.)
2   Press the Reset Switch once to restart the Call Server or wait for the Call Server to automatically reboot and the patch will be sent to the Call Server.

Alternatively, Telnet on to the Call Server via Root (as described in the Root Access to a Call Server section from page 252) and enter poweroff to reboot the Call Server. This is useful if you wish to do an upgrade remotely.

The patch will be automatically deleted from the ftp directory if the file has been accepted.

Upgrading the Call Server

The relevant new software patch must be sent to the Call Server as described above. The Call Server will be automatically upgraded during the reboot.

The current version of software being used by the Call Server can be viewed in Manager by selecting Modules, then the relevant module. The Product Version field will display the version number being used.

Upgrading the 5100 Call Server via the USB port

A USB memory stick can be used to upgrade the software on a 5100 Call Server.

1   Place the cs5100.root.tar.gz or cs5100.patch.tar.gz files on to the memory stick
2   Put the memory stick into the USB port on the front of the 5100 Call Server
3   Reboot the Call Server and the software will be upgraded.

Please note: An upgrade will take place on each reboot of the Call Server if the memory stick is left in the USB port.

Upgrading the Phone Module, Trunk Module and Voice Compression Module

The new software for a Phone Module, Trunk Module or VCM is supplied with the Call Server patch. This new software patch must be sent to the Call Server as described from page 258 before proceeding. Once the software patch has been sent to the Call Server or after a Factory Restart, the software on the Phone Module, Trunk Module and VCM can be upgraded, as follows.

Upgrading locally

1 Perform an orderly shut down of the module (press the Reset switch once)
2 When the SpliceCom LED is flashing press the Reset Switch 8 times (or no more than 10 times).
3 The LED will flash twice, flash twice again after 2 seconds and then turn off for 10 seconds.
4 The LAN activity LED will be solid for a few seconds as it downloads the software from the Call Server.

Please note: If you accidentally press the Reset Switch 8 times on a Call Server no action will be performed.

The current version of software being used by the module can be viewed in Manager by selecting Modules, then the relevant module. The Product Version field will display the version number being used.

Upgrading remotely via Telnet

1 Within the Command Prompt on a PC connected to the system enter

   telnet <ip address of the module>   eg telnet 192.168.0.248

2 When prompted for the Login name enter *diag*
3 When prompted for the Password enter the password for this module. By default this is *7388*. (The Password is set within the relevant Phone Module's configuration form.)
4 A hash (#) will appear, indicating that the previous entries where accepted and a command can be entered.
5 Enter *upgrade loader <IP address of Call Server>*   eg upgrade loader 192.168.0.1
6 When the hash (#) returns enter *upgrade now*
7 The module will reboot and automatically restart.

Please note: If the Phone Module is a DHCP client -
The Phone module has its operational software stored in flash memory. When upgrading this software as above the module does its usual DHCP request for an IP address BUT it also looks in the DHCP response for Option 66 "Boot Server Host Name" to find out the location of the software. The Phone Module then does a TFTP request to that server for the "potsmod.img" file. If it does not see that option within the DHCP response the module will not come back up as it has already erased the original software.

When the Call Server acts as the DHCP server it gives out the required fields. A Windows 2000 DHCP server can also give out this DHCP option either in general or to specific MAC addresses, but only if option 67 "Boot file name" is also specified. The configured file name is not important to the Phone Module so use any name, eg frog.txt.

In summary, it is important that the DHCP server has option 66 (and 67, if necessary) pointing to the Call Server otherwise the Phone module software upgrades will fail.

Upgrading the Software on the PCS 580

Before starting the software patch pcs580.tar.gz must be sent to the Call Server as described from page 258.

Upgrading locally

1. On the PCS 580 lift the handset.
2. Press the screen while rebooting the unit.
3. As soon as the message "`Release screen for boot options`" appears let go of the screen
4. The message "`Press screen before the countdown completes to erase the config`" will appear. Wait until the countdown finishes.
5. The message "`Press screen before the countdown completes to erase the code`" will appear.
6. Press the screen
7. When the message "`Code will be erased.  Please release screen to continue`" appears let go of the screen
8. The unit will be upgraded and the reboot will complete

The version of software being used by the PCS 580 can be viewed within the Phone Information screen which can be accessed by pressing the Settings icon [icon] five times.

Upgrading remotely via Telnet

You will need to know the IP address of the PCS 580 which is displayed on the handset by pressing the Settings icon [icon] five times.   This Phone Information screen will also display the current software version being used by the handset.

Alternatively, a list of the IP addresses for each IP phone can be viewed in Manager by selecting Utilities and then IP Addresses.

You will also need to know the phone's Password which is configured in Manager within the relevant PCS 580 configuration form. By default this is set to 7388.

The PCS 580 can be upgraded as follows:

1. Within the Command Prompt on a PC connected to the system enter

   telnet <ip address of PCS 580>   eg telnet 192.168.0.248

2. When prompted for the Login name enter *diag*
3. When prompted for the Password enter the password for this phone. By default this is *7388.* (The Password is set within the relevant Phone's configuration form.)
4. A hash (#) will appear, indicating that the previous entries where accepted and a command can be entered.
5. Enter *upgrade loader <IP address of Call Server>*   eg upgrade loader 192.168.0.1
6. When the hash (#) returns enter *upgrade now*

7   The PCS will reboot and automatically restart.

Please note: if you need to specify the call server that the upgrade will obtain the software from the IP address can be entered with the upgrade now command, eg upgrade now 192.168.0.50.

View the version of software via Telnet

The version of software being run on the PCS 580 can be viewed using Telnet as described above by entering:

ver /mnt2/tpcs

Upgrading a PCS 580 on a remote site

If the PCS 580 to be upgraded is based at a satellite office, for example, and connected via a VPN, for example, and if, during the upgrade, this connection went down the phone would become inoperable. The getupg command can be used instead to download the software to the phone first and then the upgrade will take place at the next reboot.

1   Telnet on to the PCS 580 as described above.
2   Enter *getupg <ip address of call server> admin <maintainer password>*

eg getupg 192.168.0.1 admin password

3   The software will be downloaded to the phone.
4   When this process has finished the following will be displayed:

```
Upgrade files have been saved.  Files will be installed at next reboot.
```

5   Reboot the PCS 580 at a convenient time and the phone will be upgraded.

Upgrading the Software on the PCS 570 and PCS 560

Before starting the software patch pcs5xx.tar.gz must be sent to the Call Server as described from page 258.

Upgrading locally

1   On the PCS 570 or 560 lift the handset.
2   Press the Speaker button while rebooting the unit.
3   The boot menu will be displayed
```
Press:
3 CONFIG
6 CANCEL
9 BOTH
# S/WARE
```
4   Press hash (#) for `S/WARE`.
5   The unit will be upgraded and the reboot will complete

The version of software being used by the PCS 570 or 560 can be viewed by selecting the Favourites ♥ button, then Settings 📷 and then Phone Information.

Upgrading remotely via Telnet

You will need to know the IP address of the PCS 570 or PCS 560 which is displayed on the handset by selecting the Favourites ♥ button, then Settings 🔧 and then Phone Information. This screen will also display the current software version being used by the handset.

Alternatively, a list of the IP addresses for each PCS 570 and PCS 560 can be viewed in Manager by selecting Utilities and then IP Addresses.

You will also need to know the phone's Password which is configured in Manager within the relevant IP Phones' configuration form. By default this is set to 7388.

The PCS 570 or PCS 560 can be upgraded as follows:

1   Within the Command Prompt on a PC connected to the system enter

    telnet <ip address of PCS 570/560>   eg telnet 192.168.0.248

2   When prompted for the Login name enter *diag*
3   When prompted for the Password enter the password for this phone. By default this is *7388.* (The Password is set within the relevant Phone's configuration form.)
4   A hash (#) will appear, indicating that the previous entries where accepted and a command can be entered.
5   Enter *upgrade loader <IP address of Call Server>*   eg upgrade loader 192.168.0.1
6   When the hash (#) returns enter *upgrade now*
7   The PCS will reboot and automatically restart.

Please note: if you need to specify the call server that the upgrade will obtain the software from the IP address can be entered with the upgrade now command, eg upgrade now 192.168.6.70.

View the version of software via Telnet

The version of software being run on the PCS 570/560 can be viewed using Telnet as described above by entering:

ver /mnt2/tpcs

Upgrading a PCS 570/560 on a remote site

If the PCS 570/560 to be upgraded is based at a satellite office, for example, and connected via a VPN, for example, and if, during the upgrade, this connection went down the phone would become inoperable. The getupg command can be used instead to download the software to the phone first and then the upgrade will take place at the next reboot.

1   Telnet on to the PCS 570/560 as described above.
2   Enter *getupg <ip address of call server> admin <maintainer password>*

    eg getupg 192.168.0.1 admin password

3   The software will be downloaded to the phone.
4   When this process has finished the following will be displayed:

```
     Upgrade files have been saved.  Files will be installed at next reboot.
```

5   Reboot the PCS 570/560 at a convenient time and the phone will be upgraded.

Upgrading the Software on the PCS 410/400

Before starting the relevant software patch must be sent to the Call Server as described from page 258. You will require the following files:

    PCS 410 – pcs410.tar.gz
    PCS 400 – this is supplied with the software patch for the Call Server.

Upgrading locally on the PCS 410

1   On the PCS 410 lift the handset.
2   Press the screen while rebooting the unit.
3   As soon as the message "`Press screen to erase all, release for options`" appears let go of the screen
4   The message "`Press screen before the bar completes to erase the config`" will appear. Wait until the load bar is complete.
5   The message "`Press screen before the bar completes to erase the code`" will appear.
6   Press the screen
7   When the message "`Code will be erased.  Please release screen to continue`" appears let go of the screen
8   The unit will be upgraded and the reboot will complete

The version of software being used by the PCS 410/400 can be viewed by pressing the More Options icon twice and then the Information icon.

Upgrading locally on the PCS 400

1   On the PCS 410/400 lift the handset.
2   Press the screen while rebooting the unit.
3   The unit will be upgraded and the reboot will complete

Please note that this method will also default the unit's configuration.

Upgrading remotely via Telnet

You will need to know the IP address of the PCS 410/400 which is displayed on the handset by pressing the More Options icon twice and then the Information icon. This screen will also display the current software version being used by the handset.

Alternatively, a list of the IP addresses for each PCS 410/400 can be viewed in Manager by selecting Utilities and then IP Addresses.

You will also need to know the phone's Password which is configured in Manager within the relevant IP Phones' configuration form. By default this is set to 7388.

The PCS 410/400 can be upgraded as follows:

1   Within the Command Prompt on a PC connected to the system enter

telnet <ip address of PCS 410/400>   eg telnet 192.168.0.248

2   When prompted for the Login name enter *diag*
3   When prompted for the Password enter the password for this phone. By default this is *7388.* (The Password is set within the relevant Phone's configuration form.)
4   A hash (#) will appear, indicating that the previous entries where accepted and a command can be entered.
5   Enter *upgrade loader <IP address of Call Server>*   eg upgrade loader 192.168.0.1
6   When the hash (#) returns enter *upgrade now*
7   The PCS will reboot and automatically restart.

View the version of software via Telnet

The version of software being run on the PCS 410/400 can be viewed using Telnet as described above by entering:

ver /mnt2/gtk_pcs/xpcs

Upgrading the Software on the PCS 100

The new software for the PCS 100 is supplied with the Call Server patch. This must be sent to the Call Server as described from page 258 before proceeding. Once the new software patch has been sent to the Call Server or after a Factory Restart, the PCS 100 can be upgraded as follows.

The version of software being used by the PCS 100 can be viewed by pressing the down arrow on the handset; the current software version will be displayed at the bottom of the screen. Press the down arrow or up arrow to return to the previous screen.

Upgrading locally

1   On the PCS 100 lift the handset.
2   Press the Speaker button while rebooting the unit.
3   The boot menu will be displayed
4   Press hash (#)
5   The unit will be upgraded and the reboot will complete

Upgrading remotely via Telnet

You will need to know the IP address of the PCS 100 which can be viewed via the following three methods:

1   During power up of the phone.
2   After power up press the down arrow on the handset, the IP address and current software version being used by the PCS 100 will be displayed on the screen. Press the down arrow or up arrow to return to the previous screen.
3   A list of the IP address for each PCS 100 can be viewed in Manager by selecting Utilities and then IP Addresses.

You will also need to know the phone's Password which is configured in Manager within the relevant IP Phones' configuration form. By default this is set to 7388.

1 Within the Command Prompt on a PC connected to the system enter:

telnet <ip address of PCS 100>   eg telnet 192.168.0.247

2 When prompted for the Login name enter *diag*
3 When prompted for the Password enter the password for this phone. By default this is *7388.* (The Password is set within the relevant Phone's configuration form.)
4 A hash (#) will appear, indicating that the previous entries where accepted and a command can be entered
5 Enter *upgrade loader <IP address of Call Server>*   eg upgrade loader 192.168.0.1
6 When the hash (#) returns enter *upgrade now*
7 The PCS 100 will reboot. When the PCS 100 is ready press the down arrow to check the version number to ensure the upgrade has taken place.

View the version of software via Telnet

The version of software being run on the PCS 100 can be viewed using Telnet as described above by entering:

ver /Splicecom/xpcs

Automatic Upgrade

The upgrade of any connected Phone Module, Trunk Module, VCM, PCS 580, PCS 570, PCS 560, PCS 410/400 and PCS 100 as described above will take place automatically at 4 am each day. At this time a php script will be run and any module or PCS 580/570/560/410/400/100 not running up-to-date software will be upgraded.   This will take place on each module or PCS 580/570/560/410/400/100 that the Call Server can connect to at 4 am. If the software version being run is the same as the version available on the Call Server an upgrade does not take place.

This script can also be run manually at any time of the day as follows:

1 Telnet on to the system via Root as described in the Root Access to a Call Server section on page 252
2 At the prompt enter *php –n /WebManager/upgrade.php*
3 Details of the upgrades to each connected unit, if required, will be listed.

If no upgrade is required on any module or PCS 580/570/560/410/400/100 connected the "Nothing to do" message will be displayed. To force an upgrade to each connected module or PCS 570/560/410/400/100 enter *php –n /WebManager/upgrade.php -f*

To set the time on each connected module or PCS 580/570/560/410/400/100 to match the Call Server enter *php –n /WebManager/upgrade.php –t.* This command will be run automatically at 4 am each Monday.

To view the result of an upgrade of the system via the upgrade.php script enter the following commands after logging in as described in step 1 above.

cd /logs
ls –l
cat upgrade.log

Upgrading the PCS 60 software

New PCS 60 software will be supplied in a compressed folder called, for example, PCS60.3.2.31.msi.zip or PCS60_MAC.3.2.67.pkg.zip. Before installing the new software uninstall the current version of PCS 60 running on the PC. It is not necessary to remove any short cuts or aliases created as these can be used with the new software as long as it is installed into the same location.   The installation of the new software will be the same as a new install as described in the PCS 60 section from page 58 except that if the PCS 60 is running as an IP Phone a User will already be assigned to this PC and any Preferences settings will have been saved. If the PCS 60 is running as a partner again any Preferences settings will be saved and can be used with the new software.

Upgrading the PCS 50 software

New PCS 50 software will be supplied as a file such as PCS50.2.2.652.setup.exe or PCS50.2.2.667.pkg.zip. The installation of the new software will be the same as a new install as described in the PCS 50 section from page 69. It is not necessary to remove the existing software before installation. If the PCS 50 is running as an IP Phone a User will already be assigned to this PC and any Preferences settings will have been saved. If the PCS 50 is running as a partner again any Preferences settings will be saved and can be used with the new software.

Initial Configuration Boot Menu

Each PCS 580, 570, 560, 410 and 100 has a boot menu that allows you to take the configuration of the handset back to default and/or upgrade the handset. The PCS 400 also has this facility however a menu of options is not given and the upgrade and default of the configuration is automatic.

The boot menu can be accessed as follows:

PCS 580 – lift the handset, press the screen while rebooting the handset
PCS 570/560 – lift the handset, press the Speaker button while rebooting the handset
PCS 410 – lift the handset, press the screen while rebooting the handset
PCS 100 – press the Speaker button while rebooting handset

The boot menu will be displayed giving you the following options:

PCS 580

"`Release screen for boot options`" = let go of the screen to view the following options.

"`Press screen before the countdown completes to erase the code`" = either,
- press the screen to erase any configuration settings such as a static IP address and return the handset to the default configuration whereby the unit will obtain its IP set up via DHCP, or
- wait for the countdown to complete to view the next option.

Please note that if this option is selected the phone must be able to communicate with a DHCP server to obtain its IP address. The phone will be inoperable until it can do so.

"`Press screen before the countdown completes to erase the code`" = either,
- press the screen to remove the current software and replace it with the version stored on the Call Server, or
- wait for the countdown to complete and the reboot will continue

Please note if the phone cannot find any new software the phone will reboot using the current software.

PCS 570/560/100

3 Config = press 3 to erase any configuration settings such as a static IP address and return the handset to the default configuration whereby the unit will obtain its IP set up via DHCP.

6 Cancel = press 6 to cancel the boot menu and continue with the reboot

9 Both = press 9 to erase the configuration of the handset and upgrade the software

S/ware = press hash (#) to remove the software currently being used on the handset and use the version stored on the Call Server.

PCS 410

"`Press screen to erase all, release for options`" = either,
- continue to press the screen if you wish to erase the configuration of the handset, and remove the current software and replace it with the version stored on the Call Server, or
- let go of the screen to view the following options.

"`Press screen before the bar completes to erase the config`" = either,
- press the screen to erase any configuration settings such as a static IP address and return the handset to the default configuration whereby the unit will obtain its IP set up via DHCP, or
- wait for the load bar to complete to view the next option.

"`Press screen before the bar completes to erase the code`" = either,
- press the screen to remove the current software and replace it with the version stored on the Call Server, or
- wait for the load bar to complete and the reboot will continue

PCS 400

1. Lift the handset
2. Press the screen while rebooting the unit
3. The configuration of the handset will be automatically defaulted, and the software will be removed and replaced by the version stored on the Call Server.

Accessing the log files

When requesting support from SpliceCom the Support staff may request copies of files providing a trace of the system. These files are accessed using Telnet via Root as described in the Root Access to a Call

Server section on page 252. Please then refer to the Using Telnet section from page 270 for further details.

The files that are likely to be requested are:

Admin_xxxx.log – Call set up trace
Gateway_xxxx.log – ISDN trace
Voicemail_xxxx.log – voicemail trace
Core.xxxx – trace dumps

These files can be found in the root and should be copied to the ftp directory using the following command:

mv a b   = move a to b   eg mv Admin_1265.log /ftp

Once placed in the ftp directory these files can be accessed as described in the Admin Access to a Call Server section on page 252.

You may also be asked to access the following files:

cat /dev/emlog0   catalogue file – this is main unit trace
cat /dev/emlog1   gateway trace
cat /dev/emlog2   voicemail trace
cat /dev/emlog3   phone module trace

Using Setnet

Setnet is a facility that can be used to change the network configuration of a module, such as a Phone Module, or PCS 580, 570, 560, 410, 400 or 100. Typical use would be to give a Phone Module or PCS IP phone a static IP address.
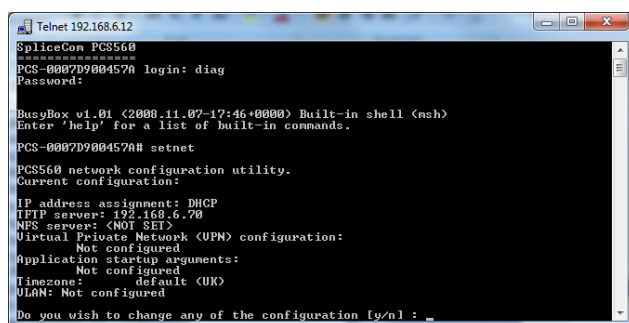
Setnet is accessed as follows:

1   Within a Command prompt enter

telnet <ip address of Phone Module/PCS 580/570/560/410/400/100>   eg telnet 192.168.0.250

then press Enter

2   You will be prompted for a Username, enter *diag*, then press Enter

3   You will be prompted for the password for this Phone Module or PCS IP phone. By default this is *7388*. (The Password is set within the relevant Phone Module or PCS IP phone's configuration form.) Then press Enter.

4   Wait for a # to appear to confirm a connection has been made to the Phone Module or PCS IP phone.

5   Enter *setnet* then press Enter

6   The current configuration will be displayed. You will be asked if you wish to change any of this configuration.

7   Enter *y* then press Enter

8   At the following prompts make any changes you require otherwise press Enter to leave the option unchanged.

9   You will be asked if you wish to save the changes. Enter *y* then press Enter

10   Telnet will display that the changes are being saved and *Done* will appear once finished.

11   Enter *poweroff* then press Enter and the Phone Module/PCS Phone will reboot

Please note:
- When you are asked if you wish to save the configuration, if you wish to cancel the changes and start again enter *n* and press Enter, or if you wish to cancel the changes and quit Setnet enter *q* and press Enter
- At any prompt you can enter a *?* for help on that option or *q* to quit.

Further information on using Setnet can be found within the Connecting a Module section from page 39 and Connecting a PCS IP Phone section from page 51.

Using Telnet

The Linux operating system used by the **max**imiser system can be directly accessed by using the "Telnet" protocol. Telnet provides simple access to the command line interface that you will need to use when troubleshooting the system. To telnet on to a call server please refer to the Root Access to a Call Server section from page 252. You will enter the directory /root from where you can enter the following commands.

Linux Commands

The following commands can be used via Telnet:

```
cd                      - change directory
cd /                    - go to root
cd /directory name      - change to the named directory eg cd /ftp
cd ..                   - back one level
ls                      - view files and directories stored in this directory
ls -l                   - view files and directories stored in this directory plus details eg size, date
ls -l B*                - view files beginning with B
pwd                     - view current location (Print Working Directory)
exit                    - exit Telnet
```

cp a b   - copy a to b   eg cp Admin_1265.log /ftp

mv a b - move a to b   eg mv Admin_1265.log /ftp
(This command can also be used to rename a file eg mv Admin_1265.log trace1.log)

rm   - remove eg rm Admin_1265.log /ftp

Where a long file name must be entered the Tab key on your keyboard can be used to auto complete a long entry. Enter the first unique letters of the file name and then press Tab, eg mv A then Tab may find mv Admin_. If only part of the file name is displayed you can continue to add unique parts of the file name and then press Tab until the complete file name is displayed.

date   - show date
date 91109002002   - set date (mmddhhmmyyyy)
/sbin/hwclock –w –u   - write time to chip

uptime   - the amount of time since the last reboot
poweroff   - poweroff and restart, so can be done remotely
reboot   - reboot and restart, SpliceCom LED flashes for a minute before rebooting

chmod   - change file permissions e.g. chmod +x frog

Up arrow on keyboard – up through history of commands
Down arrow on keyboard – down through history of commands

Setnet   - network configuration utility. Please refer to the Using Setnet section from page 269 for further details.

Chain telnets e.g. telnet into 192.168.0.1 then into 192.168.0.250

Vi Commands

vi is a file editor programme

First open a file, eg vi /SpliceCom/startVpn

i   - go into insert mode
dd   - delete line
<esc>:q!   - quit edit
<esc>:wq   - write quit

For a more extensive list of the commands available and for information on using these commands please refer to the "Linux for **max**imiser – An Introduction" document available on the SpliceCom website.

Field Descriptions

Manage Company

To view the list of Companies created on the database. Select the Company required to administer its Contacts database, Meet Me Conferences and Auto Attendant; and to view and administer the Users and Departments assigned to the Company.

Users

General Page

Name
64 Domain Characters. Name of the user.

Description
128 Characters. Description of the user.

Telephone Number
64 Digits. Primary telephone number for the user. This is the extension number of the User on the system.

Initial Phone
ID (GUID) of the initial phone this user is associated with. If blank then the user does not have an associated phone.

Locale
3 Characters. This is used to control display language for this user. If blank then user assumes hierarchical locale (local Call Server Locale or System Locale).

Company
The name (GUID) of the Company to which this User has been assigned.

Current Home
The name (GUID) of the Call Server hosting the User's voicemail.

Auto URL
128 Characters. Used with a PCS 60, 50 or 410/400. The address of the web page to be displayed on a PCS when a call involving this User is made. This can be a web page accessed via the Internet, stored on the company's Intranet or on the **max**imiser's internal web server. Used in conjunction with the No Action, On Ring and On Connect Auto URL Mode options.

Auto URL Direction
Both   The web page or directory will be displayed on both PCSs involved in an internal call to or from this User. The User's PCS will also display the web page or directory when an external call is either made or received by the User.
In   The web page or directory will only be displayed on the PCS receiving the internal call either to or from this User. The User's PCS will also display the web page or directory when an external call is received by the User.

Out   The web page or directory will only be displayed on the PCS making the internal call either to or from this User. The User's PCS will also display the web page or directory when an external call is made by the User.

Auto URL Mode

No Action   The web page specified in the Auto URL field can be viewed via the Information icon on a PCS involved in a call to or from this User

On Ring   The web page specified in the Auto URL field is automatically displayed on a PCS involved in a call to or from this User while the call is ringing

On Connect   The web page specified in the Auto URL field is automatically displayed on a PCS involved in a call to or from this User once the call is answered

Favourites   The user's Favourites (Speed Dials) are automatically displayed on a PCS involved in a call to or from this User once the call is answered

Users   The Users Directory is automatically displayed on a PCS involved in a call to or from this User once the call is answered

Departments   The Department Directory is automatically displayed on a PCS involved in a call to or from this User once the call is answered

User Class

Normal   All configuration options on a PCS 60, PCS 50, PCS 100, PCS 410/400 or PCS 560/570/580 are available.

Restricted   PCS 60 and PCS 50 - only the Speed Dials and Ring Tunes tabs of the User Configuration/Preferences form are displayed for this User.
PCS 100 – the Forwarding and DND options are removed
PCS 410/400 - only the Speed Dials, Ring Tunes and Calibration tabs of the User Configuration form are displayed.
PCS 560/570 – only the Logout, Favourites, Groups and Phone Information options are available within the Settings Screen
PCS 580 – only the Group membership and Speed Dials screens are available when pressing the Settings icon.
Manual call recording is disabled. The Record icon/button/key on a PCS 60, PCS 50, PCS 100, PCS 410/400 or PCS 560/570 is unavailable.

HotelRoom   For future use. However, if selected this will be the same as selecting Locked.

Locked   PCS 60 and PCS 50 - the User Configuration/Preferences form is not available for this User.
PCS 100 – same as Restricted
PCS 410/400 – same as Restricted
PCS 560/570 – the Settings option is not available for this User
PCS 580 – the Settings options are not available for this User
Manual call recording is disabled. The Record icon/button/key on a PCS 60, PCS 50, PCS 100, PCS 410/400 or PCS 560/570 is unavailable.

User Display Preference

Normal/Small/Large. Changes the size of the font used by the PCS 410/400.

Contact Mode

Add/Update   (Default) The User can create new Contacts and amend existing Contacts via their PCS 60, 50, 580 or 410/400

Update   The User will be able to amend existing Contacts via their PCS 60, 50, 580 or 410/400 but will not be able to create new Contacts.

No Change   The User will not be able create a new Contact or amend an existing Contact via their PCS 60, 50, 580 or 410/400.

Edit Mode
Enables or disables the User's rights to change his personal numbers and Description field via a PCS 580/570/560/410/400/60/50. The default setting, All, will enable this setting. Selecting No Personal Numbers will disable this facility.

Ex Directory
Enable/Disable. When enabled this User will not be displayed in the User directory on a PCS 580/570/560, PCS 410/400, PCS 100, PCS 60 or PCS 50.

Disable System LNR
Enable/Disable. When ticked the User's Last Number Redial list is stored locally, rather than on the call server, and will be lost when the phone is rebooted.

Details page

Home Telephone Number
64 Characters. User's home telephone number. No spaces or punctuation should be used. This information is used by voicemail, the User Directory and is matched against incoming CLI.

Mobile Telephone Number
64 Characters. User's mobile telephone number. No spaces or punctuation should be used. This information is used by voicemail, the User Directory and is matched against incoming CLI.

Spare1 Telephone Number
64 Characters. Additional telephone number. No spaces or punctuation should be used. This information is used by voicemail, the User Directory and is matched against incoming CLI.

Spare2 Telephone Number
64 Characters. Additional telephone number. No spaces or punctuation should be used. This information is used by voicemail, the User Directory and is matched against incoming CLI.

Email Address
128 Characters. Storage field for User's email address

Home Page
128 Character URL. Determines the web page to displayed as this User's Home Page on their PCS 410/400, PCS 60 and PCS 50.

External Reference
To enable a third party application to search for Users with this reference.

Telephony page

Follow Me
None   Calls will not be forwarded
Personal   All calls to the User's extension number and DDI will be forwarded

Dual Personal	All calls to the User's extension number and DDI will be forwarded, and the User's extension will ring. (Calls to the User's Department(s) will ring on the User's extension.)
All	All calls to the User's extension number and DDI will be forwarded plus any calls to a Department to which the User is a member.
Dual All	All calls to the User's extension number and DDI will be forwarded plus any calls to a Department to which the User is a member, and the User's extension will ring.
Personal External	All external calls to the User's extension number and DDI will be forwarded
Dual Personal External	All external calls to the User's extension number and DDI will be forwarded, and the User's extension will ring. (Calls to the User's Department(s) will ring on the User's extension.)
All External	All external calls to the User's extension number and DDI will be forwarded plus any calls to a Department to which the User is a member
Dual All External	All external calls to the User's extension number and DDI will be forwarded plus any calls to a Department to which the User is a member, and the User's extension will ring
Personal Internal	All internal calls to the User's extension number and DDI will be forwarded
Dual Personal Internal	All internal calls to the User's extension number and DDI will be forwarded, and the User's extension will ring. (Calls to the User's Department(s) will ring on the User's extension.)
All Internal	All internal calls to the User's extension number and DDI will be forwarded plus any calls to a Department to which the User is a member
Dual All Internal	All internal calls to the User's extension number and DDI will be forwarded plus any calls to a Department to which the User is a member, and the User's extension will ring

Follow Me To
The telephone number to redirect ALL calls to when Follow Me is set. This can be an external number or internal extension number.

Forward On Busy

None	This facility is disabled
Internal&External	All calls to the User's extension number and DDI will be forwarded to the Forward On Busy To number
Internal	Internal calls to the User's extension number and DDI will be forwarded to the Forward On Busy To number. External calls will be forwarded to voicemail, if enabled, or will receive the busy tone
External	External calls to the User's extension number and DDI will be forwarded to the Forward On Busy To number. Internal calls will be forwarded to voicemail, if enabled, or will receive the busy tone

Forward On Busy To
The telephone number to redirect calls to when Forward on Busy is set. This can be an external number or internal extension number.

Forward On No Answer

None	This facility is disabled
Internal&External	All calls to the User's extension number and DDI will be forwarded to the Forward On No Answer To number after the User's No Answer Time

Internal   Internal calls to the User's extension number and DDI will be forwarded to the Forward On No Answer To number after the User's No Answer Time. External calls will be forwarded to voicemail, if enabled, after the User's No Answer Time.

External   External calls to the User's extension number and DDI will be forwarded to the Forward On No Answer To number after the User's No Answer Time. Internal calls will be forwarded to voicemail, if enabled, after the User's No Answer Time.

Forward On No Answer To

The telephone number to redirect calls to when Forward on No Answer is set. This can be an external number or internal extension number.

Call Waiting

Controls the use of the Call Waiting facility for the User.

Disable   (default) Disables this facility

Enable   Allows the User to receive a second call and then toggle between the two calls. The User will be informed of a call waiting by an intermittent beep in the headset and via their PCS 580/570/560, PCS 410/400, PCS 100, PCS 60 or PCS 50.

PhoneTools   For use with PCS 60/50 as a partner. Enables Call Waiting but the intermittent beep in the headset is disabled. The beep is provided by the PCS 60/50.

NoBeep   Enables Call Waiting but the intermittent beep is disabled. The User will be informed of a call waiting on the screen of their PCS 580/570/560, PCS 410/400, PCS 100, PCS 60 or PCS 50.

Outgoing Call Bar

Enable/Disable. When enabled the User cannot make external calls.

Require Account Coding

Enable/Disable. When enabled the User must enter an Account Code before making an external call.

No Answer Time

The amount of time (in seconds) the User's extension will ring before it is considered not answered. (Default = 20 seconds)

Dial Plan

Name (GUID) of the Dial Plan used by the User.

Park Slot Base

Number. For use with the PCS. The value to be added to the existing Park Slot numbers accessed via the Park icons.

Park Slot Offset

Number. For use with the PCS. Specifies from which existing Park Slot number the Park Slot Base value will be added.

Login Access Code

8 Digits. The access code required to be entered to "login" as this user.

Login Idle Time

If a User has been manually logged in, then this is the amount of time (in seconds) the user can be idle before they are automatically logged out. 0 = never log out. Default = 18000 seconds.

Remote Working Code
For use with the PCS 60. Authorisation code to enable Remote PSTN. This must be the same as the Partner Login Code for the phone that this User is assigned to.

Supervisor of
For future use.

DND page

Out Of Office
Message to be displayed on a PCS 50, 60, 100, 410/400, 580/570/560 when a call is made to this User. In Office (disabled), In Meeting, Lunch (displayed as On Lunch), On Holiday, Off Site. The option selected will activate the relevant voicemail greeting, eg the In Meeting voicemail greeting, At Lunch voicemail greeting and so on.

Out Of Office Message
64 Characters. Additional text that will be displayed with the message selected above, eg until 29 September, in Boardroom.

Do Not Disturb
Enable/Disable. When enabled the User will not receive any calls and the extension is considered busy unless the incoming number matches an entry in the DND Exceptions list.

Exception Telephone Number
List of telephone numbers that will get through even when Do Not Disturb is set. With Left hand partial matching.

Capability page

Capability
16 Characters. Name of capability, eg French.

Capability %
Percentage 1-100. Strength of this capability, eg 50%.

Tunes page

Telephone Number
Used in conjunction with the CallerID and Destination Ring Tune Type. Determines for which number the ring tune will be used.

Ring Type
The instance when the selected tune is used

System	ANY system call (Telephone Number field is ignored)
Internal	ANY internal number (Telephone Number field is ignored)
External	ANY external number (Telephone Number field is ignored)
CallerID	Left hand matches the incoming Caller ID. (This has priority over all other entries).
Destination	Left hand matches the destination number of the call. (This has priority over Internal or External entries).

Parallel   Used in conjunction with Forward/Follow Me Both Personal feature. Determines the
                  ring tune to be played on the User's extension. (Telephone Number field is ignored)

Ring Tune
The Ring Tune/ringing method to be used.

Speed Dial page

Telephone Number
64 Digits. Telephone number to be dialled when this speed dial is selected.

Short Code
8 Digits. Digits to be entered to dial the Telephone Number.

Description
64 Character. Description of this speed dial.

Voicemail page

Voicemail Enabled
Enable/Disable. If enabled, callers will automatically transfer to this User's voicemail if the User's
extension is busy or not answered.

Voicemail Access Code
8 digits. Authorisation passcode to secure access to the User's voicemail.

Voicemail Assistant Telephone Number
The number (internal or external) to which callers will be transferred if they press 0 during or after the
User's voicemail greeting.

Department Voicemail 1-8
For use with a PCS 580/570/560, 410/400, 100, 60 and 50. These fields specify the Departments for which
voicemail messages can be accessed via the User's PCS. Missed and answered calls to these
Departments will also be listed.

Record Mode
Defines the mode that the call recording is to follow.

None   Automatic call recording will not take place.
Resource   Records all calls provided the appropriate voicemail resource is available at that time.
Mandatory   Records all calls.
Random   Records a random number of calls.
Trunk   Records all calls but the User's Record icon will not be activated and therefore the User
                  will not be able to cancel the recording.
Disable   Recording prohibited.

SMS Enabled
Enable/Disable. When enabled allows the User to send and receive SMS text messages via the PCS
410/400, PCS 60 and PCS 50 applications.

SMS Originator
Enter the telephone number to which you want text replies sent to, if any.

Email Address
For IMAP integration for email forwarding. The email address to receive the User's messages.

Email Server
For IMAP integration for email forwarding. The IP address of the email server supporting this facility.

Email Username
For IMAP integration for email forwarding. The name for the account on the email server.

EMail Password
For IMAP integration for email forwarding. The password for the above account

Email Mailbox
For IMAP integration for email forwarding. The folder within the email account above to receive the messages eg Inbox

Email Storage Mode
For IMAP integration for email forwarding.
Voicemail – messages remain on the voicemail server and are linked to via the email.
IMAP – messages are sent with the email and deleted from the voicemail server

Email Sync Mode
For IMAP integration for email forwarding.
Email – to synchronize voicemail to IMAP and IMAP to voicemail (used for text to speech)
Voice – synchronize voicemail to IMAP only

Voicemail Contacts

Telephone number
64 characters.
- When used with Telephone Contact Type and the Access Contact Method – the telephone number to match against the incoming CLI to allow the User to access voicemail remotely without having to enter the Access Code.
- When used with Telephone Contact Type, the Deliver and Deliver New Contact Method – the telephone number dialed by the system to allow voicemail to deliver a new message.
- When used with Telephone Contact Type and the Prompt Contact Method – the User's extension number.
- When used with the Email Contact Type – the email address where email messages informing the User of a new message will be sent.
- When used with SMS Contact Type – the email address where SMS alerts informing the User of a new voicemail message will be sent.
- When used with the Lamp Contact Type – the extension number of the Index handset across the DPNSS link
- When used with the Archive Contact Type this field is ignored, however, there must be at least a single character entered.
- When used with Archive Email Contact Type – the email address where the new call recording will be sent.

Type

Telephone	1	When accessing voicemail remotely if the incoming CLI matches the entry in the Telephone Number field the User will not need to enter their Access Code. The Contact Method field must be set to Access
	2	Voicemail will ring the number entered in the Telephone Number field to attempt to deliver a new message. The Contact Method field must be set to Deliver or Deliver New.
Email		An email message will be sent to the email address specified in the Telephone Number field when a new message is received
SMS		An SMS alert will be sent to the email address specified in the Telephone Number field when a new voicemail message is received
Lamp		Used over a DPNSS link to an Index to light the message-waiting lamp on the Index handset.
Archive		Enables an external application – such as SpliceRecord – to collect call recordings, made manually and/or automatically, from this User's voicemail box. VM Contact Method should be set to "Deliver and Delete" or "Deliver." The former is strongly recommended as it minimizes the hard disc space consumed by call recording.
ArchiveEmail		A new call recording will be sent to the email address specified in the Telephone Number field.
SMS Web		To enable SMS notification via a web page
Custom Email		*To be advised*

Method

None		Used in conjunction with Email Contact Type. The User can manually select to forward a voicemail message to the email address entered in the Telephone Number field.
Alert	1	When used in conjunction with the Email Contact Type the User will be automatically sent an email message to the email address entered in the Telephone Number field to inform them that they have received a new voicemail message
	2	When used in conjunction with the SMS Contact Type the SMS alert will be sent to the email address specified in the Telephone Number field
	3	When used in conjunction with the Lamp Contact Type the message waiting lamp will be lit on the extension number entered in the Telephone Number field
Deliver	1	When used in conjunction with the Email and ArchiveEmail Contact Type a new voicemail message will be automatically sent to the email address entered in the Telephone Number field. The message will be marked as old on the voicemail server and deleted after 7 days.
	2	When used in conjunction with the Telephone Contact Type voicemail will attempt to deliver a new voicemail message to the number entered in the Telephone Number field
Copy		Used in conjunction with the Email Contact Type. A new voicemail message will be automatically sent to the email address entered in the Telephone Number field. The message will still be considered a new message on the voicemail server
Access		Used in conjunction with the Telephone Contact Type to allow User's to access their voicemail remotely without entering the Access Code.
Prompt		Used in conjunction with the Telephone Contact Type to prompt a User for their Access Code when accessing voicemail via a Dial Plan entry.

Deliver and Delete   Used in conjunction with the Email and ArchiveEmail Contact Type. A new voicemail message will be automatically sent to the email address entered in the Telephone Number field. The message will be immediately deleted from the voicemail server.

Deliver New   Used in conjunction with the Telephone Contact Type. Voicemail will attempt to deliver a new voicemail message to the number entered in the Telephone Number field however the message will not be preceded by the usual statement indicating the number of new messages received.

Deliver Link   Used in conjunction with the Email Contact Type. A new voicemail message will be automatically sent to the email address entered in the Telephone Number field and linked to the embedded media player on the recipient's PC.

Licences page

Operator Console Licence
None   No licence has been assigned to this User and therefore Operator Console Mode is disabled

Yes   Assigns an OperatorConsole licence to this User and enables Operator Console Mode on a PCS 580/570/560/60 or the use of the PCS Operator Console application.

PCS Partner only   For use with a PCS 60 partnering a PCS 5xx IP phone. Assigns an OperatorConsole licence to this User and enables Operator Console Mode on the PCS 60 only.

Virtual User Licence
Enable/Disable. Assigns an IPVirtualUser licence to this User. To be enabled when a new User is created that will not be assigned to a physical phone and is only be used to provide PBX functionality..

Message Box Licence
Enable/Disable. Assigns a MessageBox licence to this User. Required to enable voicemail functionality for this User.

Vision Mobility Licence
For use with Vision. Please refer to Vision Installation and Configuration Manual for further details.

Vision Agent Licence
For use with Vision. Please refer to Vision Installation and Configuration Manual for further details.

OCS Licence
Enable/Disable. To enable the use of the interface, written by the Spider Group, between the **max**imiser and Microsoft Office Communications Server.

Disabled (lack of licence)
Enable/Disable. This field will be enabled when there are insufficient IPUser licences to support this User account. This field will also be enabled for Users assigned to analogue Ports 9-16 on the 5100 Call Server. Once the relevant POTS licences have been installed this field can be disabled.

Departments

General page

Name
64 Domain Characters. Name of the Department.

Description
128 Characters. Text description of the Department.

Telephone Number
8 Digits. Extension number for the Department

Out of Hours Mode

Not Used	The Out of Hours Mode is not used and the Department is fully operational
Timed	Uses the In Hours Time plan to automatically determine the operational hours of the Department.
Out Of Hours	Permanently places the Department into out of hours. The Department considers itself to be outside of its operational hours regardless of the time of day.
In Hours	Permanently places the Department into operational hours. The Department considers itself to be inside of its operational hours regardless of the time of day.

In Hours Time Plan
The name (GUID) of the Time Plan that will determine the operational hours of the Department. Any period outside the Time Plan is considered as Out Of Hours.

Auto URL
128 Characters. Used with a PCS 60, 50 or 410/400. The address of the web page to be displayed on a PCS when a call is made to and/or received by a Department. This can be a web page accessed via the Internet, stored on the company's Intranet or on the **max**imiser's internal web server. Used in conjunction with the No Action, On Ring and On Connect Auto URL Mode options.

Auto URL Mode

No Action	The web page specified in the Auto URL field is not automatically displayed on a PCS when a call is made to and/or received for the Department. However the web page can be viewed via the Information icon on the handset.
On Ring	The web page specified in the Auto URL field is displayed on a PCS while a call for this Department is ringing
On Connect	The web page specified in the Auto URL field is displayed on a PCS when a call for the Department is answered
Favourites	The user's Favourites (Speed Dials) are displayed on a PCS when a call for this Department is answered
Users	The Users Directory is displayed on a PCS when a call for this Department is answered
Departments	The Department Directory is displayed on a PCS when a call for this Department is answered

Auto URL Direction

Both	The web page or directory will be displayed on the PCS making the call and on the PCS receiving the call for this Department
In	The web page or directory will only be displayed on the PCS receiving the call for this Department

Out   The web page or directory will only be displayed on the PCS making the call to this Department

Company
The name (GUID) of the Company to which this Department has been assigned.

Current Home
The name of the Call Server hosting the Department's voicemail.

Our Account Ref
The account code to be logged against calls to this Department.

Ex Directory
Enable/Disable. When enabled this Department will not be displayed in the Departments directory on a PCS 580/570/560, 410/400, 100, 60 or 50.

Disable System LNR
To be advised.

Distribution page

Distribution Group
Name (GUID) of Group to which calls will be distributed (delivered).

Distribution Mode

All	All members of the Distribution Group will be presented with a call simultaneously
Rotary	A call is presented to the User with the highest Order number in the Distribution Group, then to the User with the next highest Order number and so on.
Sequential	Same as Rotary however subsequent calls will be presented to the next User in the Distribution Group after the User who was last presented with a call.
Manual	Members of the Distribution Group manually request a call for Department. For use with third party call centre applications that control when calls will be presented.
Blank	No call distribution will take place. For use when the Distribution Group field is empty.

Distribution Skip on No Targets
Enable/Disable. When enabled a call will move straight to the Alternate Distribution Group if all the members of the Distribution Group are busy, the Max Ring Time Before Alternate Distribution field will be ignored. Only used with the Rotary and Sequential Distribution modes.

Alternate Distribution Group
Name (GUID) of the Group to which call will be delivered if the Max Ring Time Before Alternate Distribution has expired and the call has not been answered by a member of the Distribution Group.

Alternate Distribution Mode

All	All members of the Alternate Distribution Group will be presented with a call simultaneously
Rotary	A call is presented to the User with the highest Order number in the Alternate Distribution Group, then to the User with the next highest Order number and so on.
Sequential	Same as Rotary however subsequent calls will be presented to the next User in the Alternate Distribution Group after the User who was last presented with a call.
Manual	Members of the Alternate Distribution Group manually request a call for Department. For use with third part call centre applications that control when calls will be presented.

Blank   No call distribution will take place. For use when the Alternative Distribution Group field is empty.

Alternate Distribution Skip on No Targets
Enable/Disable. When enabled a call will move straight to the Alternate Distribution Group 2 if all the members of the Alternate Distribution Group are busy, the Max Ring Time Before Alternate Distribution 2 field will be ignored. Only used with the Rotary and Sequential Alternate Distribution modes.

Alternate Distribution Group 2
Name of the Group to which a call will be delivered if the Max Ring Time Before Alternate Distribution has expired and the call has not been answered by a member of the Alternate Distribution Group.

Alternate Distribution Mode 2
All   All members of the Alternate Distribution Group 2 will be presented with a call simultaneously
Rotary   A call is presented to the User with the highest Order number in the Alternate Distribution Group 2, then to the User with the next highest Order number and so on.
Sequential   Same as Rotary however subsequent calls will be presented to the next User in the Alternate Distribution Group 2 after the User who was last presented with a call.
Manual   Members of the Alternate Distribution Group 2 manually request a call for Department. For use with third part call centre applications that control when calls will be presented.
Blank   No call distribution will take place. For use when the Alternate Distribution Group 2 field is empty.

Out of Hours Distribution Group
Name (GUID) of Group to which calls will be delivered when the Out of Hours Mode is selected or during the hours outside of the In Hours Time Plan.

Out of Hours Distribution Mode
All   All members of the Out of Hours Distribution Group will be presented with a call simultaneously
Rotary   A call is presented to the first User listed in the Out of Hours Distribution Group, then to the next in the list and so on.
Sequential   A call is presented to the next User in the Out of Hours Distribution Group after the User who was last presented with a call.
Manual   Members of the Out of Hours Distribution Group manually request a call for Department. For use with third part call centre applications that control when calls will be presented.
Blank   No call distribution will take place. For use when the Out of Hours Distribution Group field is empty.

Telephony page

Max Ring Before Announcement Time
The amount of time (in seconds) a call to the Department will ring before the comfort announcements are played.

Repeat Announcement Time.
The amount of time (seconds) between each comfort announcement. The call will be placed on hold.

Max Ring Before Alternate Distribution

The amount of time (in seconds) a call will ring the Distribution Group before being passed to the Alternate Distribution Group. Please note that 1) a call will pass to voicemail after the Max No Answer Time Before Voicemail, if set, regardless of the amount of time entered in this field, 2) this field will be ignored if Distribution Skip On No Targets is selected and all members of the Distribution Group are busy.

Max Ring Before Alternate Distribution 2

The amount of time (in seconds) a call will ring the Alternate Distribution Group before being passed to the Alternate Distribution Group 2. Please note that 1) a call will pass to voicemail after the Max No Answer Time Before Voicemail, if set, regardless of the amount of time entered in this field, 2) this field will be ignored if Alternate Distribution Skip On No Targets is selected and all members of the Alternate Distribution Group are busy.

Max Number of Active Calls Before Busy

The number of calls to be controlled by the Department. An active call is a connected call, a call waiting in the queue or a call being distributed. When this number has been exceeded any subsequent calls will be given busy or presented to voicemail.

Wrap Up Time

Default = 2 seconds. The amount of time (in seconds) given at the end of each Department call. The User will be unable to receive any further calls within this time. This will allow time for any administrative tasks at the end of a Department call. The number entered must be 2 or greater.

No Answer Time

The amount of time (in seconds) a call will ring on each extension before moving to the next member of the Distribution Group. Only used with Rotary and Sequential Distribution Modes. (Please note that a call will pass to voicemail after the Max No Answer Time Before Voicemail, if set, regardless of the amount of time entered in this field.)

Max No Answer Time Before Voicemail

The amount of time (in seconds) a call will ring for the Department before being transferred to voicemail.

Out of Hours Max No Answer Time Before Voicemail

The amount of time (in seconds) a call will ring the Out of Hours Distribution Group before being passed to voicemail.

Call Waiting

Controls the use of the Call Waiting facility for the Department.

Disable	(default) Disables this facility
Enable	Allows a User receiving a call for this Department to receive a second call and then toggle between the two calls. The User will be informed of a call waiting by an intermittent beep in the headset and via their PCS 580/570/560, 410/400, 100, 60 or 50
PhoneTools	For use with PCS 60 and 50 as a partner. Enables Call Waiting but the intermittent beep in the headset is disabled. The beep is provided by the PCS 60/50.
NoBeep	Enables Call Waiting but the intermittent beep is disabled. The User will be informed of a call waiting on the screen of their PCS 580/570/560, PCS 410/400, PCS 100, PCS 60 or PCS 50.

Voicemail page

Voicemail Enabled
Enable/Disable. If enabled the voicemail facility is available to the Department. However calls will not route to voicemail unless the Max No Answer Time Before Voicemail is set.

Message Box Mode
Determines how voicemail messages for this Department are listened to when two or more Users simultaneously accesses this Department's voicemail.

Normal   Users will be able to listen and handle all messages at the same time

Queued   Users will not listen to the same message simultaneously. First User to access this Department's voicemail will listen to message 1, then the next User will listen to message 2 and so on.

Max Message Time (Minutes)
Default = 5 minutes. Specifies the amount of time, in minutes, a caller will have to leave a message for this Department. Zero (0) can be entered.

Voicemail Access Code
8 digits. Authorisation pass code to secure the Department's voicemail.

Voicemail Assistant Telephone Number
8 digits. The extension number to which callers will be transferred if they press 0 during or after the Department's voicemail greeting.

Voicemail Target
Defines one of the following:
1   The name of the AutoAttendant to which calls will be routed, rather than voicemail, if the call is not answered, or
2   the voicemail box into which messages should be left if different from the default. Entered as !LeaveVoicemail:*extension number*, or
3   the name of the wav file to be played to a caller rather than going to voicemail. Entered as !Play:*wav file name*. The file must be stored in the Admin area of the Call Server.

Voicemail Announcement
VXML script to be run while a call is in the queue.

Record Rate
Defines the number of calls to randomly record (eg a setting of 5 will record 1 in 5 calls)

Record Group
The group of Users who will be subject to call recording

Record Mode
Defines the mode that the recording is to follow.

None   Automatic call recording will not take place.
Resource   Records all calls provided the appropriate voicemail resource is available at that time.
Mandatory   Records all calls.
Random   Records a random number of calls as defined by the Record Rate.

Trunk	Records all calls but the User's Record icon will not be activated and therefore the User will not be able to cancel the recording.
Disable	Recording prohibited.

Voicemail Contacts

Telephone number
64 characters
- When used with Telephone Contact Type and the Access Contact Method – the telephone number to match against the incoming CLI to allow the User to access voicemail remotely without having to enter the Access Code.
- When used with Telephone Contact Type, the Deliver and Deliver New Contact Method – the telephone number dialed by the system to allow voicemail to deliver a new message.
- When used with Telephone Contact Type and the Prompt Contact Method – the User's extension number.
- When used with the Email Contact Type – the email address where email messages informing the User of a new message will be sent.
- When used with SMS Contact Type – the email address where SMS alerts informing the User of a new voicemail message will be sent.
- When used with the Lamp Contact Type – the extension number of the Index handset across the DPNSS link
- When used with the Archive Contact Type this field is ignored, however, there must be at least a single character entered.

Type

Telephone	1	When accessing voicemail remotely if the incoming CLI matches the entry in the Telephone Number field the Access Code will not be required. The Contact Method field must be set to Access.
	2	Voicemail will ring the number entered in the Telephone Number field to attempt to delivery a new message. The Contact Method field must be set to Deliver.
Email		An email message will be sent to the email address specified in the Telephone Number field when a new message is received
SMS		An SMS alert will be sent to the email address specified in the Telephone Number field when a new voicemail message is received
Lamp		Used over a DPNSS link to an Index to light the message waiting lamp on the Index handset.
Archive		Enables an external application – such as SpliceRecord – to collect call recordings, made manually and/or automatically, from this User's voicemail box. VM Contact Method should be set to "Deliver and Delete" or "Deliver." The former is strongly recommended as it minimizes the hard disc space consumed by call recording.
Archive Email		A new call recording will be sent to the email address specified in the Telephone Number field.
SMS Web		For future use
Custom Email		To be advised

Method

Blank	Used in conjunction with Email Contact Type. A user can manually select to forward a voicemail message to the email address entered in the Telephone Number field.

Alert	1	When used in conjunction with the Email Contact Type an email message will be automatically sent to the email address entered in the Telephone Number field to inform the recipient that a new voicemail message has been received for the Department
	2	When used in conjunction with the SMS Contact Type the SMS alert will be sent to the email address specified in the Telephone Number field
	3	When used in conjunction with the Lamp Contact Type the message waiting lamp will be lit on the extension number entered in the Telephone Number field
Deliver	1	When used in conjunction with the Email or ArchiveEmail Contact Type a new voicemail message will be automatically sent to the email address entered in the Telephone Number field. The message will be marked as old on the voicemail server and deleted after 7 days.
	2	When used in conjunction with the Telephone Contact Type voicemail will attempt to deliver a new voicemail message to the number entered in the Telephone Number field
Copy		Used in conjunction with the Email Contact Type. A new voicemail message will be automatically sent to the email address entered in the Telephone Number field. The message will still be considered a new message on the voicemail server
Access		Used in conjunction with the Telephone Contact Type. An Access Code will not be required when accessing voicemail remotely.
Prompt		Used in conjunction with the Telephone Contact Type to prompt a User, whose extension number has been entered in the Telephone Number field, for the Department's Access Code when accessing voicemail via a Dial Plan entry
Deliver and Delete		Used in conjunction with the Email or ArchiveEmail Contact Type. A new voicemail message will be automatically sent to the email address entered in the Telephone Number field. The message will be immediately deleted from the voicemail server.
Deliver New		Used in conjunction with the Telephone Contact Type. Voicemail will attempt to deliver a new voicemail message to the number entered in the Telephone Number field however the message will not be preceded by the usual statement indicating the number of new messages received.
Deliver Link		Used in conjunction with the Email Contact Type. A new voicemail message will be automatically sent to the email address entered in the Telephone Number field and linked to the embedded media player on the recipient's PC.

Licences page

Messagebox Licence
Enable/Disable. Assigns a MessageBox licence to this Department. Required to activate voicemail for this Department.

Groups

Name
64 Domain Characters. Name of the Group.

Description
128 Characters. Text description of the Group.

Company

The name (GUID) of the Company to which this Group has been assigned.

Order

The order in which calls will be presented to the Members. The member with the higher number will be presented with a call first, then the next highest and so on.

Member

The list of Users, Phones and/or Groups that are members of this Group.

Capability

16 characters. Name of the capability.

Capability Percent

Percentage 1-100. Minimum strength of this capability.

Time Plans

Name

64 Domain Characters. Name of the Time Plan

Description

128 Characters. Text description of the Time plan

Time Zone

Enter the Time Zone required for this Time Plan, eg GB, UTC, GMT, Eire, etc

Mode

Timed	The Time Plan will use the Time Band(s) configured
Out of Hours	The Time Plan considers itself to be outside of the hours specified by the Time Band(s) regardless of the day or time
In Hours	The Time Plan is operational, in other words it considers itself to be inside the hours specified by the Time Band(s) regardless of the day or time

Alternate Time Plan

The name (GUID) of the Time Plan to be used on the days listed under Date Exception Entries.

Company

The name (GUID) of the Company to which this Time Plan has been assigned.

Time Plan Entries

Start Day of Week

The day of the week that the Time Plan is to start. Sunday-Saturday

Start Time

The time each day the Time Plan is to start. 24hr format eg 0600. This is an inclusive time.

End Day of Week

The day of the week that the Time Plan is to end. Sunday-Saturday. This entry must be a day after the Start Day of Week or the same day.

End Time

The time each day that the Time Plan is to end. 24hr format eg 1700. This is an inclusive time.

Date Exception Entries

Day of Month

The day (1-31) within the specified month that the Alternate Time Plan will be used.

Month

The month of the date that the Alternate Time Plan will be used.

Contacts

General Page

Name

64 Domain Characters. Name of the Contact. This name will be displayed in Caller Display when the incoming CLI is matched in this Contact

Description

128 characters. Text description of this Contact.

Company

128 characters. Text to identify the Contact's company/business

Department

128 characters. Text to identify the Contact's department within that company.

Direct Route To

64 Characters. The extension number, User or Department Name to which a call from the Contact is to be routed. Alternative the name of a WAV file to be played when a call is received from the Contact by entering *!Play:wav file name*. The WAV file must be stored in the FTP area of the Call Server.

Telephone Number

64 Characters. Telephone number. No spaces or punctuation should be used.

Mobile Telephone Number

64 Characters. Telephone number. No spaces or punctuation should be used.

Fax Telephone Number

64 Characters. Telephone number. No spaces or punctuation should be used.

Home Telephone Number

64 Characters. Telephone number. No spaces or punctuation should be used.

Spare 1 Telephone Number
64 Characters. Telephone number. No spaces or punctuation should be used.

Spare 2 Telephone Number
64 Characters. Telephone number. No spaces or punctuation should be used.

Ambiguous Telephone Number
64 Characters. Used to match an incoming CLI range. A question mark (?) is used to indicate where the number will be variable eg 01923287??? will match 01923 287200, 287203, 287220, 287530 etc.

Email Address
64 characters. Storage field for Contact's email address.

Auto URL
128 Characters. Used with the PCS 60, 50 and 410/400. The address of the web page to be displayed on a PCS when a call is made to and/or received from this Contact. This can be a web page accessed via the Internet, stored on the customer's Intranet or on the **max**imiser's internal web server. This field can be used in conjunction with the No Action, On Ring and On Connect Auto URL Mode options.

Auto URL Mode
No Action	The web page specified in the Auto URL field is not automatically displayed on a PCS when a call is made to and/or received from this Contact. However the web page can be viewed via the Information icon on the handset.
On Ring	The web page specified in the Auto URL field is displayed while a call to or from this Contact is ringing
On Connect	The web page specified in the Auto URL field is displayed on a PCS when a call to or from this Contact is answered
Favourites	The User's Favourites (Speed Dials) are displayed on a PCS when a call to or from this Contact is answered
Users	The Users Directory is displayed on a PCS when a call to or from this Contact is answered
Departments	The Department Directory is displayed on a PCS when a call to or from this Contact is answered

Auto URL Direction

Both	The web page or directory will be displayed on a PCS when a call is made to this Contact and on a PCS when a call is received from this Contact
In	The web page or directory will only be displayed on a PCS when a call is received from this Contact
Out	The web page or directory will only be displayed on a PCS when a call is made to this Contact

Ex Directory
Enable/Disable. When enabled this Contact will not be displayed in the Contacts directory on a PCS 580/570/560, 410/400, 100, 60 or 50.

Priority Boost
To be advised

Organisation page

Job Title
64 characters. Text to identify the job title of the Contact.

First Name
64 characters. Text to identify the first name of the Contact.

Last Name
64 characters. Text to identify the last name of the Contact

Street
64 characters. Text to identify the address of the Contact

Town/City
64 characters. Text to identify the town or city of the Contact's address

County
64 characters. Text to identify the county of the Contact's address

Post Code
64 characters. Text to identify the Post Code of the Contact's address

Country
30 characters. Text to identify the country of the Contact's address

Account (our ref)
The code to be registered with this Contact when making a call with an Account Code.

Account (their ref)
Allows an Account Code to be entered for this Contact. This code then appears in the Call Logging output against telephone calls made to or from this Contact.

Notes page

Date & Time
The date and time that the note was created. When a Note is created via a PCS 60, 50 or 410/400 the date and time will be automatically entered.

Originator
The name of the User that created the note. When a Note is created via a PCS 60, 50 or 410/400 the name of the originator will be automatically entered.

Comment
128 characters. Text field.

Meet Me Conference

Name
64 characters. A digit number to identify the conference

Organiser
64 characters. A digit number or text to identify the organiser of the conference

Date
The day that the conference should be active. Format must be yymmdd eg 030917

Start Time
The time that the conference should be active for participants to join. 24hr format eg 1100

End Time
The time that the conference should cease to accept new participants. 24hr format eg 1200

Pin
Personal Identification Number to be entered by all participants before they can join the conference

Listen ID
The ID number to be entered by participants when joining the conference to listen only.

Listen Pin
Personal Identification Number to be entered with the above Listen ID

Max Members
Specifies the maximum number of participants in this conference. Each participant occupies a voicemail connection.

Secure
For future use

Dial Plans

Name
64 Domain Characters. Name of the Dial Plan.

Description
128 Characters. Text description of this Dial Plan

Dial Delay Time
Milliseconds. Time allowed to expire before a number is forced to the line.

Dial Delay Count
Number. The number of characters after which a number is forced to the line.

Common Dial Plan
The name of the Dial Plan to be used in conjunction with this Dial Plan that typically contains common entries used by all Users.

Dial Plan Entries

Time Plan
The name (GUID) of a Time Plan that controls when this entry is active.

Call Server
The entry when dialled via this Call Server will have priority over an entry where the Call Server has not been specified.

Number Match
64 Characters. Digits to be matched with the digits dialled by the User. A left hand matching string.

Action
The action to be performed.

Busy – the busy tone will be presented

Call Waiting Disable – set the Call Waiting field to "Disable" for the User whose User Name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Call Waiting Enable – set the Call Waiting field to "Enable" for the User whose User Name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Call Waiting Phone Tools – set the Call Waiting field to "PhoneTools" for the User whose User Name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Cancel All Forwarding – turn off the Forward on Busy option and the Forward On No Answer option and set the Forward/Follow Me field to None for the User whose User Name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Conference – create a 3-way conference with the number (internal or external) entered in the Translate To field.

Dial – dial the digits entered in the Translate To field

Dial Account – make a call using the account code entered in the Translate To field

Dial Auto Answer – forces calls made to an IP extension to be automatically answered. For use with PCS 580/570/560, PCS 410/400, PCS 100, PCS 60 (IP phone) PCS 50 (IP phone) & PCS Operators Console (IP phone) only.

Dial Auto Conference – join the call taking place on the number (internal or external) entered in the Translate To field to create a 3-way conference. (Commonly known as "barge")

Dial Auto Listen – silently monitor the extension number entered in the Translate To field. Provides listen-only Call Monitor/Eavesdrop facilities to Managers, Administrators and Supervisors.

Dial Emergency – allows an external call to be made when Outgoing Call Bar is set

Dial Force Call Waiting – force the Call Waiting facility on the extension entered in the Translate To field when Call Waiting has not been set for this extension. User will hear the intermittent beep in the headset to indicate a second call is waiting.

Dial Force Call Waiting and Recall – transfer (unannounced) the call to the extension number entered in the Translate To field. If the extension is busy force the Call Waiting facility when Call Waiting has not been set for this extension. User will hear the intermittent beep in the headset to indicate a second call is waiting. If the call is not answered return the call, do not forward to voicemail.

Dial Hash – wait for a terminating # then dial the digits entered in the Translate To field

Dial Number Unavailable – the busy tone will be presented

Dial Priority – allow the call to take priority over calls queuing for the Department entered in the Translate To field.

Dial Recall – transfer (un-announced) the call to the extension number entered in the Translate To field. If the call is not answered return the call, do not forward to voicemail.

Dial Record – for outbound external calls only. Dial the digits entered in the Translate To field and the call will be recorded. A voicemail box call "OutboundRecordings" is created which can be accessed via a Department of the same name. If the User making the outbound external call is a member of a Company the recording will be stored in the voicemail box specified within the relevant Company's configuration form

Dial Transparent – allow call to by-pass an intra-module trunk. Used for fax calls between sites to be routed over the PSTN where there is a compressed (8 kbps G.729a) link – using a Voice Compression Module or card between sites. Does not consume a compression channel.

Dial Trunk – force a call on to a trunk

Do Not Disturb Disable – turn off the Do Not Disturb option for the User whose User Name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Do Not Disturb Enable – turn on the Do Not Disturb option for the User whose User Name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Do Not Disturb Exception Add – add to the Do Not Disturb Exception Telephone Number list the number entered in the Translate To field. This will be activated for the User on whose extension the short code was dialled.

Do Not Disturb Exception Remove – remove from the Do Not Disturb Exception Telephone Number list the number entered in the Translate To field. This will be activated for the User on whose extension the short code was dialled.

Do Not Disturb Exception Remove All – remove all numbers from the Do Not Disturb Exception Telephone Number list. This will be activated for the User on whose extension the short code was dialled.

Do Not Disturb Toggle – toggle on/off the Do Not Disturb option for the User whose User Name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Follow Me All – set Forward/Follow Me to All for the User whose User Name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Follow Me Both All – set Forward/Follow Me to BothAll for the User whose User Name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Follow Me Both Personal – set Forward/Follow Me to BothPersonal for the User whose User Name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Follow Me Disable – set Forward/Follow Me field to "None" for the User whose User Name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Follow Me Enable – set Forward/Follow Me field to "Personal" for the User whose User Name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Follow Me Set – set the Forward/Follow Me number field to the number entered in the Translate To field. This will be activated for the User on whose extension the short code was dialled.

Follow Me Toggle – toggle between "Personal" and "None" in the Forward/Follow Me field for the User whose User Name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Forward On Busy Disable – turn off the Forward on Busy option for the User whose User Name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Forward On Busy Enable – turn on the Forward On Busy option for the User whose User Name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Forward On Busy Set – set the Forward on Busy number field to the number entered in the Translate To field. This will be activated for the User on whose extension the short code was dialled.

Forward On Busy Toggle – toggle on/off the Forward on Busy option for the User whose User Name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Forward On No Answer Disable – turn off the Forward on No Answer option for the User whose User Name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Forward On No Answer Enable – turn on the Forward on No Answer option for the User whose User Name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Forward On No Answer Set – set the Forward on No Answer number field to the number entered in the Translate To field. This will be activated for the User on whose extension the short code was dialled.

Forward On No Answer Toggle – toggle on/off the Forward on No Answer option for the User whose User Name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Login – log in as the User specified by the extension number and Login Code entered in the Translate To field. The extension number and Login Code are separated by an asterisk eg 2040*1234

Logout – log out the User currently logged in and return to the User assigned to this phone

Malicious Call Trace – for Australian approvals. Send a tone on the ISDN line to the carrier to alert them of a malicious call.

Night Service In Hours – set the Mode field to "In Hours" for the Department whose Department Name or extension number has been entered in the Translate To field.

Night Service Out Of Hours – set the Mode field to "Out of Hours" for the Department whose Department Name or extension number has been entered in the Translate To field.

Night Service Timed – set the Mode field to "Timed" for the Department whose Department Name or extension number has been entered in the Translate To field.

None – no action will be performed

Out Of Office Disable – set the Out of Office field to "In The Office" for the User whose User Name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Out Of Office Enable – set the Out of Office field to "In Meeting" for the User whose User Name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Out of Office Lunch – set the Out of Office field to "Lunch" for the User whose User Name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Out of Office Holiday – set the Out of Office field to "On Holiday" for the User whose User Name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Out of Office OffSite – set the Out of Office field to "Off Site" for the User whose User Name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Page – page the phone(s) assigned to the User Name, extension number or Group name entered in the Translate To field

Park – park a call in the park slot number entered in the Translate To field. An entry of -n (where n is a number) in the Translate To field instructs the system to automatically park the call in next available slot starting at n, eg –1

Pickup – pick up a call ringing on a phone assigned to the User Name, Department Name, Group Name or extension number entered in the Translate To field. If this field is blank this Action will pick up the longest ringing call on the system.
or
- pick up a call parked in the park slot number entered in the Translate To field.

Please note that a number of two digits or less entered in the Translate To field will translate to a Park Slot number.

Relay Close – close the External Relay port (1 or 2) specified in the Translate To field

Relay Open – open the External Relay port (1 or 2) specified in the Translate To field

Relay Pulse – close the External Relay port (1 or 2) specified in the Translate To field for 5 seconds and then reopen the port

Ring Back When Free - ring back the number entered in the Translate To field when the extension becomes available. This features requires an ESP licence.

Set Event – activate the Event entered in the Translate To field

Set Home Number – set the Home Telephone number field to the number entered in the Translate To field. This will be activated for the User on whose extension the short code was dialled.

Set Mobile Number – set the Mobile number field to the number entered in the Translate To field. This will be activated for the User on whose extension the short code was dialled.

Set Spare1 Number – set the Spare1 number field to the number entered in the Translate To field. This will be activated for the User on whose extension the short code was dialled.

Set Spare2 Number – set the Spare2 number field to the number entered in the Translate To field. This will be activated for the User on whose extension the short code was dialled.

Time Plan Disable – set Out of Hours Mode for the Time Plan entered in the Translate To field.

Time Plan Enable – set In Hours Mode for the Time Plan entered in the Translate To field.

Time Plan Timed – set Timed Mode for the Time Plan entered in the Translate To field.

Voicemail Add Call Me – to create a Voicemail Contact entry for voicemail call back, to instruct the voicemail server to ring the number entered in the Translate To field when a new message has been received, for the User on whose extension the short code was dialled.

Voicemail Disable – turn off the Enable Voicemail option for the User whose User Name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Voicemail Enable – turn on the Enable Voicemail option for the User whose User Name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Voicemail Greeting – access the daily greeting for the User or Department whose name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Voicemail Greeting Permanent – access the permanent greeting for the User or Department whose name or extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Voicemail Remove All Call Me – to remove all Voicemail Contact entries for voicemail call back for the User on whose extension the short code was dialled.

Voicemail Remove Call Me – to remove a Voicemail Contact entry for voicemail call back for the number entered in the Translate To field for the User on whose extension the short code was dialled.

Voicemail Retrieve – access voicemail for the User or Department whose extension number has been entered in the Translate To field. If the Translate To field is blank this Action will be activated for the User on whose extension the short code was dialled.

Translate To
128 Characters. The digits to be dialled by the system.

Translate CLI To
128 Characters. The extension number to be matched in the relevant DDI Call Plan. The digits entered in the Number Match field of the extension number's DDI Call Plan entry will be sent as outgoing CLI.

LCR Plan
The name (GUID) of the LCR Plan to be used when an outgoing external call is made as a result of this entry.

Companies

Name
64 Domain characters. Name of the Company

Description
128 Characters. Text description of the Company.

Park Timeout
Default 5 mins. Any call parked by a User who is a member of this Company will be held in the Park Slot for the time (in seconds) specified in this field and then represented to the User.

Outbound Record Mailbox
The name of the User or Department into whose voicemail box outbound external calls will be recorded when the DialRecord Dial Plan Action is used to make an external call.

Inbound Record Mailbox
The name of the User or Department into whose voicemail box inbound external calls will be recorded when a DDI Plan entry is configured to automatically record inbound external calls for this Company.

Trunks

General page

Name
64 Doman Characters. Name of this trunk. Not configurable.

Description
128 Characters. Text description of this trunk.

Presentation Number
64 Characters. The Telephone Number to be presented as a CLI on outgoing calls when no other alternative exists.

DDI Call Plan
Name (GUID) of the DDI Plan to be used to route incoming calls on this trunk and to determine the outgoing CLI for a call.

Dial Plan
The name of the Dial Plan to be used if the DDI Call Plan field is empty. For use with ISDN equipment connected to this port.

Capacity
Number. The quantity of channels/connections available on this trunk.

Reserve Incoming Capacity
Number. The quantity of channels specified in the Capacity field, to be reserved solely for incoming calls.

Reserve Voice Capacity
For future use

Reserve Data Capacity
For future use

Port
ISDN – default 0
SIP trunk– default 5060
H323 trunk – default 1720

Out of Service

Enable/Disable. If enabled outgoing calls will not be passed to this Trunk.

ISDN page

TEI Number

Terminal Equipment Identifier - to identify each unit connected to a particular ISDN line. For Point to Point lines this is set to 0. For Point to Multi-Point line this can also be set to 0, however if multiple devices are sharing a Point to Multi-Point line it should be set to 127 and the exchange decides on the TEI's to be used by this unit.

NT mode

Enable/Disable. If enabled this Trunk will act as an S0 port.

Mode

Support for ETSI / DPNSS / Index DPNSS / PRA Lite ISDN (New Zealand)

Trace Layer 2

Enables tracing via Telnet

Trace Layer 3

Enables tracing via Telnet

Restart Mode

Not used

Restart Command

A debugging tool that provides a way of restarting channels on an ISDN trunk. It is a comma separated list of channels to restart (1-15, 17-31) or zero to restart the whole trunk interface.

DPNSS Node Number

The component number given to the Call Server. Typically 3 digits.

DPNSS Peer Node Number

The component number to which this port is connected

Outgoing Call Party Plan

Determines the numbers to be added to the outgoing CLI.

Unknown	(default) The carrier determines this option
National	Adds a 0 (zero)
International	Adds 44

National Prefix

Determines the prefix for incoming calls. When left blank, the default, 0 is used. Entering a non-digit such as "." or ">" leaves the digit space blank.

International Prefix

Determines the prefix for incoming calls. When left blank, the default, 00 is used. Entering a non-digit such as "." or ">" leaves the digit space blank.

NSI Message Waiting On String
Defines the string that must be sent across a DPNSS link in order to turn on the Message Waiting light on the handsets connected to the PBX at the remote end. (This feature is not used with the Index.)

NSI Message Waiting Off String
Defines the string that must be sent across a DPNSS link in order to turn off the Message Waiting light on the handsets connected to the PBX at the remote end. (This feature is not used with the Index.)

Channel Allocation Mode
Default / Top Down / Bottom Up
The Default option will allocate channels dynamically, alternatively the allocation can be forced from top to bottom (1-30) or bottom to top (30-1) on this trunk.

Audio Format
A-law / u-law. By default audio will be sent on this Trunks as A-law. To send audio on the Trunk as u-law select this option.

Disable Clock Source
Enable/Disable. If enabled the trunk will not attempt to obtain its clock/timing source from the equipment, such as a video conferencing device, connected to this port.

Send Progress
NT only / Yes / No
For use in Australia. Disables progress tones being sent back to a looped call interfering with the tones for the original call.

SIP page

Compression Type
None   Calls to be made without using compression
G729A8K   Use G.729a based compression supported via a Voice Compression Module or Card
Relay   Relay the media packets, allowing RTP packets to go through a NAT/firewall by using the
                   Call Server as a single address for all RTP

Allow Non-Compressed Audio
Enable/Disable. For use with G.729 Compression Type. This option will allow A-law or u-law to be selected by the remote endpoint instead of G.729.

NAT Server
IP address allocated by the ADSL provider, in xxx.xxx.xxx.xxx form

NAT Port
Port to be used by NAT = 5060

NatMode
Static   Use IP address entered above
STUN   For future use

Use TCP
Enable/Disable. Allows the use of TCP rather than UDP (default).

No Progress
Enable/Disable. If enabled this option will stop SIP PROGRESS messages being sent.

Receive on Port
Default = 5060

Registrar Address
Address provided by SIP provider

Registration User Name
User Name provided by SIP provider

Registration Password
Password provided by SIP provider

Call Server
Determines the Call Server to be used for this trunk.

H323 Page

Compression Type
None   Calls to be made without using compression
G729A8K   Use G.729a based compression supported via a Voice Compression Module or Card
Relay   Relay the media packets, allowing RTP packets to go through a NAT/firewall by using the
                Call Server as a single address for all RTP

Allow Non-Compressed Audio
Enable/Disable. For use with G.729 Compression Type. This option will allow A-law or u-law to be
selected by the remote endpoint instead of G.729.

NAT Server
IP address allocated by the ADSL provider, in xxx.xxx.xxx.xxx form

NAT Port
Port to be used by NAT = 1720

Nat Mode
Static   Use IP address entered above
STUN   For future use

Allow Fast Start
Enable/Disable. Must selected at both ends. Required for link from Call Server to Call Server.

Allow Hold Music
Enable/Disable. For use when this trunk is connected to another Call Server. If enabled, this option will
send hold music information on this trunk.

Calls Server
Determines the Call Server to be used for this trunk.

Trunk Groups

Name
64 Domain Characters. Name of the Trunk Group.

Description
128 Characters. A text description of this Trunk Group.

Trunk
List of Trunks (GUIDs) that are members of the Trunk Group

DDI Plans

Name
64 Domain Characters (GUID). Name of the DDI Plan

Description
128 Characters. Text description of this DDI Plan

DDI Entries

Order
Determines the position of an entry within a DDI Plan. DDI Plan entries are read from top to bottom by the system. Entries will be listed in sequential order with low numbers at the top and higher numbers lower down.

Time Plan
The name (GUID) of the Time Plan to determine when this entry is active.

Number Match
64 Characters. The incoming DDI number to be matched. An exact match is required. This number will also be the outgoing CLI for the extension number entered in the Translate To field.

Bearer Capability
Set to Voice, Data, Video or Fax. Informational field only.

Caller ID
64 Characters. The number to be matched against incoming CLI. Left hand matching.

Translate To
64 Characters. The number or name to which the incoming call is to be routed. This can be a User, Contact, Department, or external number. If a DDI Call Plan entry is to be used to determine the outgoing CLI for a call enter the User or Department's extension number in this field.

Locale
Not used

LCR Plans

Name
64 Domain Characters (GUID). Name of the LCR Plan.

Description
128 Characters. Text description of this LCR Plan

LCR Plan Entries

Time Plan
The name (GUID) of a Time Plan that determines when this entry is active.

Number match
64 Characters. Digits to be matched with the digits dialled. A left hand matching string.

Action
The action to be performed. For an explanation of each Action please refer to page 294.

Dial
128 Characters. The digits to be dialled by the system.

DTMF
64 Digits. The digits to be dialled once a connection has been made.

Trunk Group
Name (GUID) of the Trunk group to be used for the external call made as a result of this entry.

Action Secondary, Dial Secondary, DTMF Secondary, Trunk Group,
Same as Primary but used if the Primary does not succeed.

Timeout
Milliseconds. Time allowed to expire before a number is forced to the line. A # after the number dialled with ignore this Timeout entry.

Voicemail Ports

General Page

Name
64 Domain Characters. Name of the Voicemail Port

Description
128 Characters. Text description of this Voicemail Port

Product Version
Software version of the voicemail application. This information will only appear when this port communicates with a voicemail server.

Call Server
The name of the Call Server (GUID) on which this Voicemail Port is active. If left blank then all call servers will be used determined by the User or Departments Current Home field.

Remote Call Server
Redirects requests for voicemail services to this Call Server

Dial Plan
The name of the Dial Plan to be used by the voicemail facility. Required for Call back features, Auto Attendant and any feature where the voicemail server is dialling a number.

H.323 Address
For use when an external voicemail PC is used. Entry can be in Domain Name or IP Address format.

Public Address
The address to be used in the email announcement. Allows an external address or domain name to be specified for access over the Internet.

Capacity
Defines the maximum number of calls allowed into this Voicemail Port

Email Smart Host
The IP Address of the SMTP mail server to be used for voicemail email functionality.

Email Source Address
The name of the email account used for the voicemail email functionality. This will be the name displayed in the From field of an email message unless an entry is made in the Email Source Name field.

Email Source Name
The name of the email account to be displayed in the From field of an email message. This will ensure that Microsoft Outlook does not associate the Email Source Address with the email account of the caller leaving the message.

Auto Attendant Server
Enables and disables the ability by this voicemail port to support an Auto Attendant.

Voicemail Server
Enables and disables the ability by this voicemail port to support the voicemail facility.

Music Server
Enables and disables the ability by this voicemail port to support Music On Hold.

Queue Message Server
Enables and disables the ability by this voicemail port to support announcements for a Department.

Recording Server
Enables and disables the ability by this voicemail port to support call recording.

Conference Server
Enables and disables the ability by this voicemail port to support call conferencing.

Extended Attendant
Enable/disables the ability by this voicemail port to support Extended Attendant. Must be enabled when using VXML Scripts.

Enhanced Speech Services Capacity
Defines the maximum number of Enhanced Speech Processing (ESP) calls allowed into this Voicemail Port. The same number of ESPSession licences must be installed before configuring this field.

Internal Collect Number
Number to be included in SMS message to allow access to voicemail internally.

External Collect Number
Number to be included in SMS message to allow access to voicemail externally.

Text to Speech Voice
Enter either *David* or *Millie* to determine the voice to be used with VXML and Text-to-Speech. If left blank will use David.

Require Mailbox Access Code
Enable/Disabled. If enabled all Users will be prompted to enter their Voicemail Access Code when accessing their voicemail via a dial plan entry using the Voicemail Retrieve action.

Debug page

SMTP / IMAP / Concurrency / IMAP Poll / VXML Data
Enables tracing via Telnet in the voicemail.log

Music Channels

Name
64 characters. This field will indicate when the Music Channel will be used. If this Music Channel is to be active when a Department call is on hold enter the Department name. If the Music Channel is to be active when a specific User places a call on hold enter the User Name. If the Music Channel is to be active when calls to a specific Company are placed on hold enter the Company Name. If the Music Channel is to be active for all calls placed on hold (unless a specific User or Department or Company Music Channel exists) enter DefaultHoldMusic.

Playlist
64 characters. If the message/music is to be stored internally enter the name of the WAV file, stored in the Admin area of the Call Server, to be played. If this field is left blank the default WAV file will be played ("Please hold"). Alternatively, if the music/message is being played from an external source enter any text that does not match the name of a WAV file stored on the Call Server.

Multicast Address
18 characters. The Multicast address of the Call Server, eg 224.0.1.51

Multicast Port
128 characters. The address of the Multicast Port for the Music Channel. Where the music is being stored internally this must be an even number not already in use (not less than 1024). Where the music is being sourced externally the number entered must be equal to 16640 + (2 x port number).

Source Server
The name of the Call Server (GUID) on which the Music Channel is active. The WAV file is stored on this Call Server and calls to this Call Server will be played the music/message indicated in the Playlist field. If this field is left blank the Music Channel will be active on all Call Servers operational in the one system.

Events

Number
2 characters. The number to be used to active the Event. Any number (0-99) can be activated by a Dial Plan entry. Numbers 1 and 2 can be activated via the Relay Input ports.

URL
128 Characters. The address of the web page to be displayed on a PCS when the Event is activated. This can be a web page accessed via the Internet, stored on the customer's Intranet or on the **max**imiser's internal web server.

User Group
The name of the Group (GUID) to which the Event will be sent. Only members of this Group will receive the Event thereby ensuring only the relevant Users are sent the Event.

Phone Group
The name of the Group (GUID) to which the Event will be sent. Only members of this Group will receive the Event thereby ensuring only the relevant Users are sent the Event.

VXML Scripts

Name
Text used to identify this Script. Alpha-numeric characters only. No spaces or punctuation should be used. This will be the name of .php file created on the internal Web Server.

Time Plan
The name (GUID) of the Time Plan used to determine when the entry within the Go To field will be used. Any period outside the Time Plan is considered as out of hours.

Go To
The name of the Form to be presented to the caller when first accessing this VXML Script.

Go To Out of Hours
The name of the Form to be presented to the caller when the VXML Script is considered out of hours as specified by the Time Plan.

Server Address
IP address used when connection made via LDAP, typically 127.0.0.1, or when the VXML Script is to be run from a remote call server.

Server User
Administrator name used when connection made to the above server, typically Manager

Server Password

Administrator password used when connection made to the above server, typically managerpassword

Forms

Name

Text used to identify this Script. Alpha-numeric characters only. No spaces or punctuation should be used.

Say the following

The message to be played when this form is presented to the caller. This can be text that will be converted to speech or the name of the .wav file stored on the internal Web Server.

Wait for Digits Timeout

Default = 10 seconds. The amount of time (in seconds) after the "Say the Following" message has been played that the Form will wait for the caller to make a choice from the Menu.

Min Number of Digits to Wait for

The number of digits that the Form will wait for the caller to dial before executing an action.

Menu Items

Number Match

The digits to be dialled by the caller to activate this Menu Item. Can contain numbers 1-9, * and #. A ? entered in this field will match any number entered.

Say the following

The message to be played when this Menu Item is activated. This can be text that will be converted to speech or the name of the .wav file stored on the internal Web Server.

Delay Time after speaking

The amount of time (in seconds) after the "Say the Following" message has been played before the caller is transferred to the Transfer To number or presented with the Go To Form.

Transfer To

The extension number of the User or Department to which the caller will be transferred.

Transfer Timeout

The amount of time (in seconds) after which the Transfer To extension number is considered not answered.

Transfer Audio

Enter the name of the WAV file to be played during the transfer, if required. This file must be stored in the /Web directory on the call server.

Go To

The name of the Form to be presented to the caller on activating this Menu Item (unless the Transfer To field contains an entry).

Go To On Busy

The name of the Form to be presented to the caller when the Transfer To extension number is busy.

Say on Busy
The message to be played when the Transfer To extension number is busy. This can be text that will be converted to speech or the name of the .wav file stored on the internal Web Server.

Go To On No Answer
The name of the Form to be presented to the caller when the Transfer To extension number is considered not answered as specified by the Transfer Timeout field.

Say on No Answer
The message to be played when the Transfer To extension number is considered not answered as specified by the Transfer Timeout field. This can be text that will be converted to speech or the name of the .wav file stored on the internal Web Server.

Auto Attendant

Name
1 character. The digit to be dialled by a caller to active the required selection. This field can contain numbers 0-9, *, # or ?. The ? indicates a 3 second wait before the entry in the Telephone Number field is dialled.

Telephone Number
The extension number, User or Department name to which a caller will be transferred when making this selection.

Modules

Call Server

General page

Name
64 Domain Characters. Name of this module.

Description
Description of the module

Location
To provide location text information for SNMP

Locale
3 Characters. The locale that controls Caller Display type, ring sequences and the language used by voicemail on this particular module. If blank then assume System Locale.

Serial Number
The primary MAC address of this module.

IP Address
IP Address of this module, in xxx.xxx.xxx.xxx form.

Product Version
Software version running on this module

Module Status
Member   A module that is a part of the system
Available   A module that is waiting for permission to join
NonMember   Non members are modules that are part of another system, but because of the
                    network are visible for this domain. This mode prevents them joining the wrong
                    domain.

Flash LED
The Flash LED button will instruct the Call Server's SpliceCom LED to flash 20 times.

Call Server/Remote Call Server page

Administration Mode
Primary   A Call Server that is the master administrative module controlling the configuration
                  database.
Secondary   A Call Server that is linked to a Primary Call Server and receives its configuration from the
                  Primary.

DHCP Server Mode
Enable   The Call Server will act as the DHCP server using the IP Address range specified in the DHCP
                Start Address and DHCP End Address fields. If another DHCP Server has been detected on
                the network and the Call Server is rebooted it will keep its previous IP address and disable its
                own DHCP server.
Disable   DHCP is disabled; the Call Server will use the IP address entered in the IP Address field within
                the General page.
Client   The Call Server will act as the DHCP client obtaining an IP address from a DHCP Server on the
                network.

DHCP Start Address
First IP address of the IP address range to be used when the DHCP Server Mode is set to Enable. Default
= 192.168.0.2. (Please note that this field is irrelevant if the DHCP Server Mode is set to Disabled.)

DHCP End Address
Last IP address of the IP address range to be used when the DHCP Server Mode is set to Enable. Default
= 192.168.0.250. The system assumes a linear distance between the two numbers and has no knowledge
of sub address masks. (Please note that this field is irrelevant if the DHCP Server Mode is set to Disabled.)

DHCP Base Address
Default = 192.168.0.0. The network on which the Call Server will allocating IP addresses.

DHCP Base Mask
Default = 255.255.255.0. IP address mask to be used by the Call Server.

DHCP Router
Default Gateway

DNS Domain
Domain name

DNS Primary
DNS primary server

DNS Secondary
DNS secondary server

SMTP Server Address
The IP address of the mail server to receive the Warnings and Notifications emails for this Call Server.

Email address for Warnings
The email address to be sent a message when an entry is made in the Warnings page.

Email address for Notifications
The email address to be sent a message when a notification is made, eg when the Call Server needs to perform an "Ask Master".

Public (Internet) IP Address
The public/external IP address for the module. For use with multiple Call Servers using a public network through NAT routers.

Auto Add Phones
Enable / Disable. If enable, this feature will allow Phone Modules and IP Phones to automatically register with the Call Server. If disabled Phone Modules and IP Phones must be manually registered.

Global BLF
Enable/Disable. When enabled this feature provides a system wide Busy Lamp Field view across all networked Call Servers for all User's on the system. Requires multicast Router support to transport Global BLF info between sites. Where required, this option must be enabled on all Call Servers – not just the Primary Call Server.

Global Call Log
Enable/Disable. When enabled in a multi-Call Server network this feature provides a unified Call Logging output from all Servers to be presented on the Primary Call Server. Where required, this option must be enabled on all Call Servers – not just the Primary Call Server.

IP Routes page
This page can be used to create IP routes for destinations where the packets need to route back on to the LAN.

Intramodule Trunks page

Name
64 Domain characters. Name of this trunk.

Compression Type
None   Calls to be made without using compression
G729A8K   Use G.729a based compression supported via a Voice Compression Module or Card

Relay   Relay the media packets, allowing RTP packets to go through a NAT/firewall by using the Call Server as a single address for all RTP

Capacity
Number of connections to be allowed via this Call Server.

Reserved Incoming Capacity
The number of channels to be reserved for incoming calls.   Outgoing calls will not be made if the remaining capacity is exceeded.

IP Address (override)
Default - blank = public IP address. The IP address that the module should use as a source address when communicating with the modules listed below. Can be an internal or public IP address.

Members

Member
The module to whom compressed calls will be made.

Order
Determines the position of an entry within the Members list.

Debug page

Socket / RAS / H225 / H245 / H450 / Gatekeeper / LDAP / SIP / Database
Enables tracing via Telnet

Call Flow
Basic / Normal / Detail / Debug
Determines the amount of information displayed in the Admin (emlog0) output.

Phone Module

General Page

Name
64 Domain characters. Name of this module.

Description
Description of the module

Location
To provide location text information for SNMP

Locale
3 Characters.   The locale that controls Caller Display type, ring sequences and the language used by voicemail on this particular module. If blank then assume System Locale.

Serial Number
The primary MAC address of this module.

IP Address
IP Address of this module, in xxx.xxx.xxx.xxx form.

Product Version
Software version running on this module

Module Status

Member	A module that is a part of the system
Available	A module that is waiting for permission to join
NonMember	Non members are modules that are part of another system, but because of the network are visible for this domain. This mode prevents them joining the wrong domain.

Call Server
Name of the Call Server to which the Phone Module is registered.

Flash LED
The Flash LED button will instruct the Phone Module's SpliceCom LED to flash 20 times.

Phone Module Page

Password
The code to be used when accessing this module via Telnet. If blank, the default setting of 7388 is used.

Trunk Module

Name
64 Domain Characters. Name of this module.

Description
Description of the module

Locale
3 Characters.   The locale that controls Caller Display type, ring sequences and the language used by voicemail on this particular module. If blank then assume System Locale.

Serial Number
The primary MAC address of this module.

Module Status

Member	A module that is a part of the system
Available	A module that is waiting for permission to join
NonMember	Non members are modules that are part of another system, but because of the network are visible for this domain. This mode prevents them joining the wrong domain.

IP Address
IP Address of this module, in xxx.xxx.xxx.xxx form.

Flash LED
The Flash LED button will instruct the Trunk Module's SpliceCom LED to flash 20 times.

Product Version
Software version running on this module.

Call Server
Name of the Call Server to which the Trunk Module is registered.

External Gateway Module

Name
64 Domain Characters. Name of this module.

Description
Description of the module

Locale
3 Characters.   The locale that controls Caller Display type, ring sequences and the language used by voicemail on this particular module. If blank then assume System Locale.

Serial Number
The primary MAC address of this module.

IP Address
IP address of the remote end, in xxx.xxx.xxx.xxx form, or the name, e.g. sip.example.com.

Product Version
Not applicable

Module Status
Member	A module that is a part of the system
Available	A module that is waiting for permission to join
NonMember	Non members are modules that are part of another system, but because of the network are visible for this domain. This mode prevents them joining the wrong domain.

Call Server
Name of the Call Server to which the module is registered.

Voice Compression Module

Name
64 Domain Characters. Name of this module.

Description
Description of the module

Locale
3 Characters.   The locale that controls Caller Display type, ring sequences and the language used by voicemail on this particular module. If blank then assume System Locale.

Serial Number
The primary MAC address of this module.

Module Status

Member   A module that is a part of the system
Available   A module that is waiting for permission to join
NonMember   Non members are modules that are part of another system, but because of the network are visible for this domain. This mode prevents them joining the wrong domain.

IP Address

IP Address of this module, in xxx.xxx.xxx.xxx form.

Flash LED

The Flash LED button will instruct the VCM's SpliceCom LED to flash 20 times.

Product Version

Software version running on this module.

Call Server

Name of the Call Server to which the VCM is registered.

Phones

Analogue extension port

POTS Phone page

Name

64 Domain Characters. The name of this port.

Description

128 Characters. Text description for this port.

Location

To provide Location Text Information for SNMP

Product Version

Not applicable

Caller Display Type

None   Call Display information will not be sent to the extension and the extension will not ring once after the relevant Phone Module has been rebooted
Default   PCS 10 (suitable for most analogue handsets)
Alarm   For use with alarm systems, DTMF is not sent out but allows the alarm system to send DTMF tones back out.
Bell   USA
BT   UK
BTNoMessages For use with InterQuartz Discovery analogue phones
DTMF   Europe
Fax   For use with a fax machine connected to this port, provides a strong/robust ring tone improving the fax machine's ability to answer a call
FaxServer   For Fax Server support, please refer to page 182 for further details.
Lamp   For use with a PCS 505 or PCS 5 to support the message waiting lamp

Interquartz   For use with InterQuartz Gemini analogue phones
Pay Phone   For use with a pay phone
PCS 10   For use with the PCS 10
PCS 10 Lamp   For use with a PCS 10 to support the message waiting lamp
System Phone   For use with the PCS 520, ensures CLI information is displayed correctly and enables
                   paging and auto answer functionality
Trunk   Sends a disconnect signal to the end device.

Caller Display Length
The number of characters to be sent to the port. Default = 0 which represents 20 characters.

Partner Login Code
1-8 digit code. Used to secure ability to run PCS50 as an assistant on this port.

Offhook Working
Enable/Disable. Allows the use of headset without the need to lift the receiver

Regenerate DTMF
Enable/Disable. If enabled DTMF will be regenerated on this port rather than passed Inband.

DTMF Receive Mode
Auto / H.245 Signal / RTP Event / Inband only
Default = Auto. Used when equipment such a fax server or alarm system is connected to this port.
Determines how the analogue port will receive DTMF tones from the line. Leave as Auto unless specified
by the third party equipment.

Unlicensed Port
Enable/Disable. This field is used in conjunction with a 5100 Call Server. The first eight analogue ports on
a 5100 Call Server are activated on installation. However ports 9-16 require a POTS licences to enable
each port. This field will be ticked when the phone port is unlicensed. The tick should be removed when
a POTS licence has been purchased.

Page Port
None   For use if this port is to be used as a standard extension with analogue phone.
Paging   For use if powered speakers are to be connected to this port for use as a Tannoy.
MusicOnHold   For use if an external Music-on-Hold source is to be connected to this port.

Remote PSTN
The number to be dialled by the system when the User assigned to this port is working from home or on
a mobile and wishes to handle calls as though they were in the office.

Remote Idle
Seconds. The number of seconds that the User's home telephone will remain idle before a call from the
maximiser system is cleared.

User
The name of the User to whom this port is assigned.

IP Phone

Fields specific for an IP Phone listed under a Call Server:

Name
64 Domain Characters. The name of this phone.

Description
128 Characters. Text description for this phone.

Location
To provide Location Text Information for SNMP

Password
The code to be used when accessing this IP Phone via Telnet. By default the code is 7388 but this can be changed via this field.

Product Version
The version of software running on this IP Phone.

Model
Unspecified – setting for PCS 100, 560, 570 and 580
Mono 480 x 320
Colour 480 x 320
Colour 640 x 480 – setting for PCS 410/400

Serial Number
MAC address of this phone.

Partner Login Code
1-8 digit code. Used to secure ability to run a PCS 60 or PCS 50 as a partner on this port.

Remote PSTN
The number to be dialled by the system when the User assigned to this phone is working from home or on a mobile and wishes to handle calls as though they were in the office.

Remote Idle
Seconds. The number of seconds that the User's home telephone will remain idle before a call from the **max**imiser system is cleared.

Headset Working
Enable/Disable. Allows the use of headset without the need to lift the receiver

Screen Saver Mode
For use with the PCS 580, 570, 560, 410, 400

Dim on Idle	Not available. If selected this will be the same as selecting Bright Always.
Blank on Idle	The screen saver will appear after the handset has been idle for the number of seconds indicated in the Screen Saver Params field.
Blank at Night Dim on Idle	Not available. If selected this will be the same as selecting Blank at Night.
Blank at Night	The Screen Saver will appear between the times entered in the Screen Saver Params field
Bright always	The screen saver will not be used.

Screen Save Params
Enter as Idle Timeout (Secs),Night Start Time,Night End Time
1. If the Screen Save Mode field is set to Blank on Idle enter the number of seconds that the handset is to remain idle before the screen saver will appear, eg 900,0,0, or
2. If the Screen Save Mode field is set to Blank at Night enter the time in 24 hour clock that the screen saver will appear followed by the time in 24 hour clock that the screen saver will be removed eg 0,1730,0800

User
The name of the User to whom this phone is assigned.

Unassigned Phones

Name
64 Domain Characters. Name of this phone.

Type of Phone
Name of the type of unit requesting registration

Serial Number
MAC address of this phone.

Status
Member — A phone that is permitted to register on the system
Available — A phone that is waiting for permission to join
NonMember — A phone that is not permitted to register on the system

Call Server
Name of the Call Server to which the phone is connected.

Unassigned Modules

Please refer to the field list for each Module from page 299.

System

System Name
64 Domain Characters. Name to identify the system. When licences are used this field must match exactly the name provided with the licence keys.

Supplier
Name to identify the supplier of the system. This field must be used when licences are configured and must match exactly the name provided with the licence keys.

Description
128 Characters. Text description of the system.

Locale
3 Characters. As per Windows 2000 locales, eg eng for English UK. Controls default Caller Display type, ring sequences, language used by voicemail.

Call Logging Password
64 Characters. Password used to gain access to Call Logging information. (Default=none = no access).

Diagnostic Engineering Password
64 Characters. Password used to gain access to Root. (Default=password)

Web Password
64 characters. Password to gain access to the internal Web Server. (Default = password)

Maintainer Password
64 characters. Password to gain access to the Admin and voicemail areas. (Default = password)

Licences

Key
34 character string

Qty
Number of licences purchased

Licence
Name of the licence type

Administrators

Name
64 Domain Characters. Name of the administrator.

Description
128 Characters. Text description of the administrator.

Password
64 Characters. Password for this administrator.

Show Enabled
Enabled/Disabled. If ticked enables the use of the Show tick boxes below. If not ticked the Administrator has full access to the database dependent on the rights and access allowed.

Show Users, Departments, Groups etc
Allows the Administrator to view and amend the selected section of the database.

Add/Delete Users, Departments, Groups etc
Allows the Administrator to add a new entry and delete an existing entry within the selected section of the database

Company

Determines the Company that this Administrator can manage. The Administrator will only be able to access the entries within the selected sections of the database that relate to the selected Company. If this field is blank the Administrator will have access to all entries within the selected sections of the database.

Default Page

Determines the page that will be displayed when Manager is first opened. If this list box is blank the default page will be the Users list.

Authorised Access

A list of the Administrators given access to the database.

Rights

A list of the Rights given to this Administrator

Utilities

Backup Now

Creates a backup of the configuration stored on the database and is accessed via the Admin area.

Ask Master

Used when connecting a Secondary Call Server to a Primary Call Server. The IP Address of the Call Server is entered in the Ask Master field. This address will automatically revert back to 192.168.0.2 to ensure this process is not activated in error.

Merge Remote System

Used when connecting a Secondary Call Server to a Primary Call Server. The two configurations will merge rather than the secondary defaulting and downloading the configuration from the Primary. This feature will retain as much of the configuration from the secondary as possible, therefore the configuration will, for example, contain two Dial Plans - Standard and Standard-1, two Administrators - Manager and Manager-1, and so on. The User's extension numbers within the configuration on the secondary will be renumbered.

Set Time

To set the time used by the system.

Call Status

List of calls currently active on the system.

Voicemail Status

Lists all voicemail boxes and the number of unplayed and total voicemail messages for each account.

File Import/Export

This screen provides the ability to import and export Area Codes, Contacts, Dial Plan, DDI Call Plans and Users.

Modify Users

Allows User parameters to be modified off line using an Excel spreadsheet.

Warnings
This screen provides a list of each reboot of the system and licence error warnings.

IP Addresses
Lists IP addresses against associated name and MAC address.

Mailbox Licences
Displays the number of MessageBox licences installed, the number allocated and which Users and Department they have been allocated to.

Sorted Lists
Sorts by Name, Extension & Initial Phone and searches for duplicate entries. Also lists by physical Phone.

User Status
Checks that User locations are consistent across all active Call Servers.

Voicemail GUIDs
List of Users, Departments and Companies with their corresponding GUIDs (global unique identifier). Assists with identification when the GUID is displayed rather than the name, for example, when searching for a User's voicemail folder on the voicemail server.

Custom Script
This link will open custom.php. This feature can be used to create a custom web page to be viewed via Manager, for example, displaying which Users have DND set. The page to be displayed must be name custom.php and copied on to the web server prior to selecting this link.

Setup Total Control
For use with Total Control. Full instructions will be given on the purchase of this service.

Area Codes

Name
64 characters. The name to be displayed in Caller Display when the incoming CLI is matched by the Ambiguous Telephone Number field.

Ambiguous Telephone Number
64 Characters. Used to match an incoming CLI range. A question mark (?) is used to indicate where the number will be variable eg 01923?????? will match 01923 287200, 01923 678938 etc.

WAN Links

General page

Name
Unique name for this link using a maximum of 8 alpha characters. Do not use any numbers or punctuation.

Description
Text description for this link.

Account Name Out
Account name sent to distant end to ask for access

Account Password Out
Account password sent to distant end to ask for access

Account Name In
Name presented from distant end to be checked with password for access authentication.

Account Password In
Password presented from distant end to be checked with name for access authentication.

Telephone Number
ISDN number to dial to reach the distant end. This field is only required at the end that makes the call.

DDI Telephone Number

Blank	No ISDN access into this unit
*	ISDN access allowed via any data call on any ISDN port
Any number	ISDN access allowed when matched to this DDI on any ISDN port

Time Plan
The name (GUID) of the Time Plan that will determine the operational hours of the link, if left blank the link will work 24x7

Dial Usage Quota
Not used

IP Header Compression
IP headers within PPP will be compressed.

Caller IDs
This link will only answer if the call originates from one of the CLIs listed here.

Trunks

Trunks on this module on which outgoing calls will be made. Please note: Data calls are not routed via Dial Plan entries or LCR Plan entries so enter any required prefixes in the Telephone Number field.

IP Routes page

IP Address
The TCP/IP address of the distant network or host

IP Mask
The subnet mask for the IP Address above

IP Gateway
Gateway to be used

Metric

A value from 0 to 15 where 0 is the best value. If in doubt set it as 1, equates to values in RIP.

PPP page

Multilink

Enable/Disable. Enables the negotiation of Multilink. Some products abandon a call/link if asked to negotiate. Typically set to Enable

Maximum Reconstructed Reception Unit

Specifies the maximum number of octets in the Information fields of reassembled packets, typically set to 1500

Maximum Receive Unit

The negotiated packet size that can be received from a Multi Link PPP connection typically set to 1500

Sequence Numbers

Short (2 bytes) or Long (4 bytes). To negotiate whether short or long sequence numbers will be allowed for a packet. Recommend Short is used to reduce the overhead on the system.

Authentication Protocol

PAP   Password Authentication Protocol, provides a simple method for the peer to establish its identity using a 2-way handshake. This is done only upon initial link establishment

Chap   Challenge-Handshake Authentication Protocol is used to periodically verify the identity of the peer using a 3-way handshake, and therefore is more secure. This is done upon initial link establishment, and MAY be repeated anytime after the link has been established

Authentication Method

Incoming   Authentication is only verified in one direction.

Outgoing   Authentication is only verified in one direction

Bothway   Authentication is verified in both directions.

On Demand page

Min. Call Time

Specifies the amount of time the call should connected before closing through inactivity

No Session Idle Period

Specifies the time period a link needs to be idle before the Call Server will close the connection

Active Session Idle Period

The amount of time (in seconds) the connection will stay up when a session is open across the link (eg Telnet, browser to far end is open) but there is no data transfer (ie the open session is idle).

Idle Traffic Check

Not used

Min. Channels Required

Specifies the number of ISDN Channels the Call Server should use on initial dial up

Max. Channels Allowed

Specifies the total number of ISDN Channels the Call Server will use for this link.

Extra Bandwidth Threshold

Percentage. Specifies the amount of throughput on the existing link before an extra ISDN channel is added.

Reduce Bandwidth Threshold

Percentage. Specifies the level of throughput on the existing link before an ISDN channel is removed

Incoming Traffic Keeps Link Up

Enable/Disable. If this field is enabled incoming traffic will be considered when timing the idle period ie the connection will be open and not idle.

Technical Data

Approval

These products are CE compliant according to the European Directives for Telecommunications, Safety and EMC.

Safety Caution

The telephone ports on the Phone43xx, Remote Call Server, 5100, 5108, 5315 and 5330 are classified as TNV-2 ports. These ports may generate voltages exceeding normal safe (SELV) limits, e.g. when ringing. Care should be exercised when using these ports (avoid touching the contacts) and only TNV-2 or TNV-3 compliant devices may be attached.

The ISDN ports on the Call Server, Trunk, Remote Call Server, 5100 and 5108 units are classified as TNV-1 ports. This means that the connecting device (e.g. ISDN line from the telephone company) normally operates within safe limits but can on occasion have high Voltage transients. The unit will not be harmed by these transients but care should be exercised when handling the cable connected to the telephone company's NTU, i.e. avoid touching the cable contacts.

All other ports operate within safe (SELV) limits. For continued safe operation these ports may only be connected to other SELV ports.

EMC Caution

Warnings
These products are Class A. In a domestic environment these products may cause radio interference in which case the user may be required to take adequate measures.

General

Module Dimensions (mm)	487(w) x 224(d) x 43(h)
5100 Call Server Weight (kg)	3.5
5108 Call Server Weight (kg)	1.7
5315 Phone module Weight (kg)	3.3
5330 Phone module Weight (kg)	3.3
4100 Call Server Weight (Kg)	3.6
4200 Trunk module Weight (Kg)	3.4
43xx Phone module Weight (Kg)	3.5
4140 Remote Call Server Weight (Kg)	3.7
4400 Voice Compression module Weight (Kg)	3.4
Environmental	0 to 40°C
	85% Relative Humidity, non condensing

AC power supplies

The lump in line PSUs used with the products operate from a 230Vac 50Hz mains supply.

Port Definitions

Call Server 5100

DC Power Inlet	2.5mm jack socket for 48V, 1.35A power supply.
POTS ports (16)	Telephone ports – see safety statement concerning TNV-2.
	Off hook voltage = 48V
	Ringing Voltage = 44Vrms
	Loop current = 25mA
LAN ports (4)	10/100BaseT Full Duplex, Auto MDI/MDI-X LAN ports.
	These ports may also provide power over LAN to IEEE 802.3af compliant attached devices.
LAN LINK port	10/100BaseT Full Duplex, Auto MDI/MDI-X LAN port.
BRI ports (4)	Basic rate ISDN ports meeting CTR3 – see safety statement.
PRI ports (2)	Primary rate ISDN ports meeting CTR4 – see safety statement.
Event inputs (2)	Opto-coupled inputs to trigger events – see text for details.
Relay outputs (2)	Opto-coupled outputs to control external devices – see text for details.
USB Host Interface	USB 2 (High/Full and Low speeds)

Phone 53xx

DC Power Inlet	2.5mm jack socket for 48V, 1.35A power supply.
POTS ports (xx)	Telephone ports – see safety statement concerning TNV-2.
	Off hook voltage = 48V
	Ringing Voltage = 44Vrms
	Loop current = 25mA
LAN ports (4)	10/100BaseT Full Duplex, Auto MDI/MDI-X LAN ports.
	These ports may also provide power over LAN to IEEE 802.3af compliant attached devices.
LAN LINK port	10/100BaseT Full Duplex, Auto MDI/MDI-X LAN port.
Event inputs (2)	Opto-coupled inputs to trigger events – see text for details.
Relay outputs (2)	Opto-coupled outputs to control external devices – see text for details.
USB Host Interface	USB 2 (High/Full and Low speeds)

Call Server 5108

DC Power Inlet	2.5mm jack socket for 48V, 1.35A power supply.
POTS ports (4)	Telephone ports – see safety statement concerning TNV-2.
	Off hook voltage = 48V
	Ringing Voltage = 44Vrms
	Loop current = 25mA
LAN ports (4)	10/100BaseT Full Duplex, Auto MDI/MDI-X LAN ports.
	These ports may also provide power over LAN to IEEE 802.3af compliant attached devices.
LAN LINK port	10/100BaseT Full Duplex, Auto MDI/MDI-X LAN port.
BRI ports (2)	Basic rate ISDN ports meeting CTR3 – see safety statement.
Event input	Opto-coupled input to trigger events – see text for details.
Relay output	Opto-coupled output to control external devices – see text for details.

4100 Call Server/4200 Trunk

DC Power Inlet	2.5mm jack socket for 48V, 1A power supply.
DC Power over LAN	5 pin DIN socket for 48V, 2.7A power supply.

LAN ports (front)	10/100BaseT Full Duplex, Auto MDI/MDI-X LAN ports.
	These ports may also provide power over LAN to IEEE 802.3af compliant attached devices.
LAN port (back)	10/100BaseT Full Duplex, MDI LAN port. (Call Server only)
BRI ports	Basic rate ISDN ports meeting CTR3 – see safety statement.
PRI port	Primary rate ISDN port meeting CTR4 – see safety statement.
WAN port	V.11 (X.21) WAN interface operating up to 2.048Mb/s.
Event inputs	Opto-coupled inputs to trigger events – see text for details.
Relay outputs	Opto-coupled outputs to control external devices – see text for details.

Phone 43xx

DC Power Inlet	2.5mm jack socket for 48V, 1A power supply.
PCMCIA	Support for 5V and dual voltage 5V/3.3V cards. Support is also dependent on available s/w drivers.
LAN port	10/100BaseT Full Duplex, MDI LAN port.
POTS ports	Telephone ports – see safety statement concerning TNV-2.
	Off hook voltage = 48V
	Ringing Voltage = 44Vrms
	Loop current = 25mA

4140 Remote Call Server

DC Power Inlet	2.5mm jack socket for 48V, 1A power supply.
LAN port (back)	10/100BaseT Full Duplex, MDI LAN port.
BRI ports	Basic rate ISDN ports meeting CTR3 – see safety statement.
PRI port	Primary rate ISDN port meeting CTR4 – see safety statement.
WAN port	V.11 (X.21) WAN interface operating up to 2.048Mb/s.
Event inputs	Opto-coupled inputs to trigger events – see text for details.
Relay outputs	Opto-coupled outputs to control external devices – see text for details.
POTS ports	Telephone ports – see safety statement concerning TNV-2.
	Off hook voltage = 48V
	Ringing Voltage = 44Vrms

4400 Voice Compression Module

| DC Power Inlet | 2.5mm jack socket for 48V, 1A power supply. |
| LAN port | 10/100BaseT Full Duplex, MDI LAN port. |

Cable Specifications

Warning: The products are not designed to directly connect to cables that are routed external to a building.

RJ45 connections

The RJ45 ports on the units (i.e. ISDN ports and LAN ports) are connected to other devices using standard category 5 UTP patch cables. These may be supplied with the units but are also commercially available from numerous retail outlets.

Alternatively custom lengths using category 5 UTP cable and 8 pin RJ45 plugs can be made up as follows:

End A	Cable Notes	End B
1	Twisted	1
2	Pair	2
3	Twisted	3
6	Pair	6
4	Twisted	4
5	Pair	5
7	Twisted	7
8	Pair	8

The allocation of pairs of pins to twisted pairs is important for signal integrity, e.g. if the wire from pin 1 was not twisted with the wire from pin 2 then any LAN link made with this cable would be prone to generating errors.

X.21 WAN cable

The V.11 WAN port is connected to the telephone company's NTU (Network Termination Unit) by the cable described below. The WAN port connects to the shielded plug with UNC 4-40 locking screws and the NTU connects to the unshielded plug with M3 locking screws. This cable is available from your reseller or may be constructed as follows:
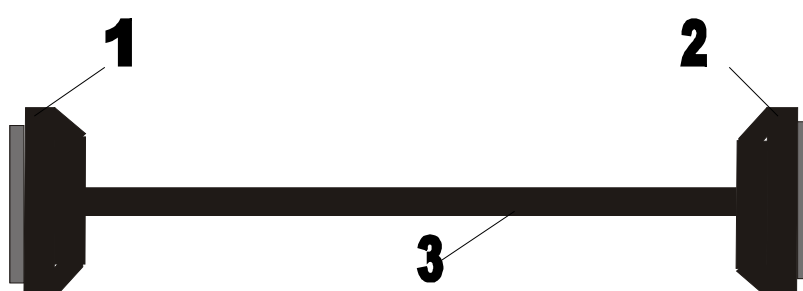
Item	Description
1	15 Way D-Type Shielded Plug with UNC 4-40 locking screws.
2	15 Way D-Type Unshielded Plug with M3 locking screws.
3	6 twisted pair screened cable

Pin Connections

End A	Name	Cable Notes	End B
11	Receive – B	Twisted	11
4	Receive – A	Pair	4
2	Transmit – A	Twisted	2
9	Transmit – B	Pair	9

3	Control – A	Twisted	3
10	Control – B	Pair	10
5	Indicate – A	Twisted	5
12	Indicate – B	Pair	12
6	SE-Timing – A	Twisted	6
13	SE-Timing B	Pair	13
8	Ground	––	8

End A hood is connected to the Screened Cable Drain Wire.
End B is not connected to the screen in any way

Relay Outputs

External devices may be switched by the relay outputs provided by the 3.5mm stereo jack sockets.

Socket   Plug
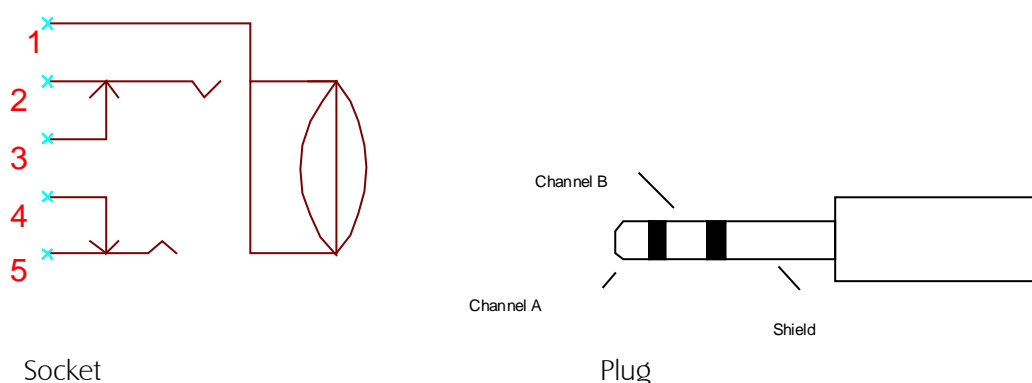
Pin   **Description**
1   Connected to unit chassis
2   Relay contact
3   Not connected
4   Not connected
5   Relay contact

The shield connector of the mating plug is connected to the unit chassis.

The on board relays are solid state optically isolated devices that can connect plug contacts "channel A" to "channel B". Their characteristics are as follows:

Switching ability = a.c. or d.c signals of either polarity.
Absolute Maximum Voltage = 60V peak. Beyond this, damage will occur.
Recommended Maximum Voltage = 48V.
Maximum relay on impedance = 15 Ohms.
Minimum relay off impedance = 500MOhms.
Recommended Maximum current = 140mA.
Current Limit = 300mA.

Event Inputs

External event inputs are provided using 4 pin mini DIN connectors that have the following pin out:
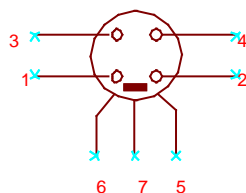
View looking into socket.

Pin	Description
1	Limited power source return
2	Signal detect -
3	Limited power source feed (5V)
4	Signal detect +
5	Chassis connection
6	Chassis connection
7	Chassis connection

Pins 5, 6 & 7 are contacts that connect the shield of the mating mini DIN plug to the unit chassis.

The input is an opto-isolator that detects current between pins 4 and 2.
It can be used in one of two modes:

External power source

In this mode the external power source is switched across pins 4 and 2 by the external event, e.g. a mechanical switch on a door entry system.

The input is isolated from the chassis and has the following specification:
Absolute Maximum Voltage = 60V. Beyond this, damage will occur.
Recommended Maximum Voltage = 48V.
Input impedance = 1000 Ohms.
Recommended on current > 3mA.
Recommended off current = 0mA.

To activate the opto-isolator pin 4 should be at a higher voltage than pin 2. No damage will occur if the voltage polarity is reversed, however no event will be detected.

Internal power source

For convenience a limited power source is provided on pins 3 and 1. This can be used to drive the opto-isolator using a switch in series to gate the current to the opto-isolator. Alternatively pin 1 can be connected to pin 2 and pin 3 to pin 4 and a switch used between pins 4 and 2 to short out the power source.

Power Source open circuit Voltage = 5V
Power Source impedance = 2000 Ohms
No damage will occur to the power source if it is shorted out.
The power source should not be used for any other purpose.

PCS 580G Technical Details

Approval

The product is CE compliant according to the European Directives for Safety and EMC.

Safety Caution

The product should not be disassembled. There are no user serviceable parts.

General

Dimensions (mm): 277(w) x 136(d) x 208(h)
Weight (Kg): 1.6
Environmental: 0 to 40°C
85% Relative Humidity, non condensing

LCD

Technology: TFT
Resolution: 800x480
Colours: 262144

Port definitions

LAN port: 10/100/1000BaseT Full Duplex, Auto MDI/MDI-X LAN port and 802.3af compliant power sink
PC port: 10/100/1000BaseT Full Duplex, Auto MDI/MDI-X LAN port.
Handset port: For connection of supplied handset.
Headset port: General 4 pin headset connection. Inner pins connect to earpiece and outer pins connect to microphone.
USB Host Interface (2 off): USB 2.0 supporting Hi-Speed (480Mb/s), Full-Speed (12Mb/s) and Low-Speed (1.5Mb/s). Support of devices also depends on driver availability.

Power Supply Considerations

The product is powered from the LAN port.
The PCS580G is a class 3 802.3af compliant powered device, i.e. draws less than 15.4W from the 802.3af power source.

Care should be exercised when adding USB devices as exceeding the available power will cause the PCS580G to shut down. Sufficient power is generally available to drive a USB keyboard and mouse. If greater power is required then an externally powered USB hub may be used.

PCS 570G Technical Details

Approval

The product is CE compliant according to the European Directives for Safety and EMC.

Safety Caution

The product should not be disassembled. There are no user serviceable parts.

General

Dimensions (mm): 277(w) x 136(d) x 208(h)
Weight (Kg): 1.5
Environmental: 0 to 40°C
85% Relative Humidity, non condensing

LCD

Technology: TFT
Resolution: 240x320
Colours: 262144

Port definitions

LAN port: 10/100/1000BaseT Full Duplex, Auto MDI/MDI-X LAN port and 802.3af compliant power sink
PC port: 10/100/1000BaseT Full Duplex, Auto MDI/MDI-X LAN port.
Handset port: For connection of supplied handset.
Headset port: General 4 pin headset connection. Inner pins connect to earpiece and outer pins connect to microphone.
USB Host Interface: USB 2.0 supporting Hi-Speed (480Mb/s), Full-Speed (12Mb/s) and Low-Speed (1.5Mb/s). Support of devices also depends on driver availability.

Power Supply Considerations

The product is powered from the LAN port.
The PCS570G is a class 2 802.3af compliant powered device, i.e. draws less than 7W from the 802.3af power source.

PCS 570/560 Technical Details

Approval

The product is CE compliant according to the European Directives for Safety and EMC.

Safety Caution

The product should not be disassembled. There are no user serviceable parts.

General

Dimensions (mm): 277(w) x 136(d) x 208(h)
Weight (Kg): 1.5
Environmental: 0 to 40°C
85% Relative Humidity, non condensing

LCD

Technology: TFT
Resolution: 240x320
Colours: 262144

Port definitions

LAN port: 10/100BaseT Full Duplex, Auto MDI/MDI-X LAN port and 802.3af compliant power sink
PC port: 10/100BaseT Full Duplex, Auto MDI/MDI-X LAN port.
Handset port: For connection of supplied handset.
Headset port: General 4 pin headset connection. Inner pins connect to earpiece and outer pins connect to microphone.

PCS570 only:
USB Host Interface: USB 2.0 supporting Hi-Speed (480Mb/s), Full-Speed (12Mb/s) and Low-Speed (1.5Mb/s). Support of devices also depends on driver availability.

Power Supply Considerations

The product is powered from the LAN port.
The PCS560 is a class 1 802.3af compliant powered device, i.e. draws less than 4W from the 802.3af power source.
The PCS570 is a class 2 802.3af compliant powered device, i.e. draws less than 7W from the 802.3af power source.

N.B. a PCS570 with no USB device plugged in will draw the same power as a PCS560 but will still identify itself as a class 2 device.

PCS 410 Technical Details

Approval

The product is CE compliant according to the European Directives for Safety and EMC.

Safety Caution

The product should not be disassembled. There are no user serviceable parts.

The internal Cold Cathode Fluorescent Lamps used to back light the LCD operate at high voltage and could pose a shock risk if exposed.

General

Dimensions (mm)	275(w) x 190(d) x 145(h)
Weight (Kg)	1.3
Environmental	0 to 40°C
	85% Relative Humidity, non condensing

LCD

Resolution   640 x 480
Colours   262144

Port definitions

USB Host Interface   USB 2 (High/Full and Low speeds)
PC port   10/100BaseT Full Duplex, Auto MDI/MDI-X LAN port.
LAN port   10/100BaseT Full Duplex, Auto MDI/MDI-X LAN port and 802.3af compliant power sink
Headset port   General 4 pin headset connection. Inner pins connect to earpiece and outer pins connect to microphone.
Handset port   For connection of supplied handset.

Power Supply Considerations

The product is powered from the LAN port.

An 802.3af compliant power supply can provide up to 12.95W to the LAN port. Care should be exercised when adding USB devices as exceeding the available power will cause the PCS410 to shut down. Sufficient power is generally available to drive a USB keyboard and mouse. If greater power is required then an externally powered USB hub may be used.

PCS 400 Technical Details

Approval

The product is CE compliant according to the European Directives for Safety and EMC.

Safety Caution

The product should not be disassembled. There are no user serviceable parts.

The internal Cold Cathode Fluorescent Lamps used to back light the LCD operate at high voltage and could pose a shock risk if exposed.

General

Dimensions (mm)   275(w) x 190(d) x 145(h)
Weight (Kg)   1.73
Environmental   0 to 40°C
                   85% Relative Humidity, non condensing

LCD

Resolution   640 x 480
Colours   262144

Port definitions

PCMCIA	Support for 5V and dual voltage 5V/3.3V cards. Support is also dependent on available s/w drivers.
USB Host Interface	USB 1.1
DC Power Inlet	2.5mm jack socket for optional 48V power supply.
PC port	10/100BaseT Full Duplex, Auto MDI/MDI-X LAN port.
LAN port	10/100BaseT Full Duplex, Auto MDI/MDI-X LAN port and 802.3af compliant power sink
Headset port	General 4 pin headset connection. Inner pins connect to earpiece and outer pins connect to microphone.
Handset port	For connection of supplied handset.

Power Supply Considerations

The product may be powered from either the LAN port or the DC Power Inlet. The DC Power Inlet will take precedence if both are supplied.

An 802.3af compliant power supply can provide up to 12.95W to the LAN port. Care should be exercised when adding USB or PCMCIA devices as exceeding the available power will cause the PCS400 to shut down. The nominal power requirement for the PCS400 alone is 11W. Sufficient power is generally available to drive a USB keyboard and mouse. If greater power is required then the DC Power Inlet should be used (or alternatively an externally powered USB hub).

The DC Power Inlet may be connected to a 48Vdc power source. (Available from your re-seller.) This will provide sufficient power for any USB or PCMCIA compliant devices.

PCS 100 Technical Details

Approval

The product is CE compliant according to the European Directives for Safety and EMC.

Safety Caution

The product should not be disassembled. There are no user serviceable parts.

General

Dimensions (mm)	275(w) x 190(d) x 145(h)
Weight (Kg)	1.1
Environmental	0 to 40°C
	85% Relative Humidity, non condensing

LCD

Resolution	64 x 128
Monochrome shades	2

Port definitions

PC port   10/100BaseT Full Duplex, Auto MDI/MDI-X LAN port.
LAN port   10/100BaseT Full Duplex, Auto MDI/MDI-X LAN port and 802.3af compliant power sink
Headset port   General 4 pin headset connection. Inner pins connect to earpiece and outer pins
                             connect to microphone.
Handset port   For connection of supplied handset.

Power Supply Considerations

The product is powered by an 802.3af compliant power source. The nominal power requirement is 3W.

PCS 520 Technical Details

Approval

The product is CE compliant according to the European Directives for Telecommunications, Safety and EMC.

Safety Caution

The product should not be disassembled. There are no user serviceable parts.
All the ports are classified as TNV-3. When the line port is connected to a telephone line, these ports may exhibit Voltages above normal safe (SELV) limits (e.g. during ringing) and may exhibit high Voltage transients. Care should be exercised when using these ports, i.e. avoid touching the contacts.

General

Dimensions (mm)   215(w) x 198(d) x 80(h)
Weight (Kg)   0.9
Environmental   0 to 40°C
                            85% Relative Humidity, non condensing

Port Definitions

Line   Socket for supplied lead to connect to a telephone line, RJ-11
Data   4 pin socket to connect to a modem/FAX, RJ-11
Headset port   4 pin socket for a headset, RJ-9
Handset port   For connection of supplied handset, RJ-9

PCS 10 Technical Details

Approval

The product is CE compliant according to the European Directives for Telecommunications, Safety and EMC.

Safety Caution

The product should not be disassembled. There are no user serviceable parts.
The product may only be used with the supplied AC power adapter.
All the ports are classified as TNV-3. When the line port is connected to a telephone line, these ports may
exhibit Voltages above normal safe (SELV) limits (e.g. during ringing) and may exhibit high Voltage
transients. Care should be exercised when using these ports, i.e. avoid touching the contacts.

General

Dimensions (mm)   215(w) x 198(d) x 80(h)
Weight (Kg)   0.9
Environmental   0 to 40°C
                       85% Relative Humidity, non condensing

AC Power Supplies

The AC adapter, required to enable the caller display functionality, operates from a 230Vac 50Hz mains
supply.

Port Definitions

Line   Socket for supplied lead to connect to a telephone line.
Data   4 pin socket to connect to a modem/FAX
Headset port   4 pin socket for a headset
Handset port   For connection of supplied handset
Power   For connection of optional AC supply

PCS 5 Technical Details

Approval

The product is CE compliant according to the European Directives for Telecommunications, Safety and
EMC.

Safety Caution

The product should not be disassembled. There are no user serviceable parts.
All the ports are classified as TNV-3. When the line port is connected to a telephone line, these ports may
exhibit Voltages above normal safe (SELV) limits (e.g. during ringing) and may exhibit high Voltage
transients. Care should be exercised when using these ports, i.e. avoid touching the contacts.

General

Dimensions (mm)   160(w) x 225(d) x 85(h)
Weight (Kg)   0.65
Environmental   0 to 40°C
                       85% Relative Humidity, non condensing

Port Definitions

Line   Socket for supplied lead to connect to telephone line.
Handset port   For connection of supplied handset
Headset port   4 pin socket for a headset, RJ-9

PCS 505 Technical Details

Approval

The product is CE compliant according to the European Directives for Telecommunications, Safety and EMC.

Safety Caution

The product should not be disassembled. There are no user serviceable parts.
All the ports are classified as TNV-3. When the line port is connected to a telephone line, these ports may exhibit Voltages above normal safe (SELV) limits (e.g. during ringing) and may exhibit high Voltage transients. Care should be exercised when using these ports, i.e. avoid touching the contacts.

General

Dimensions (mm)   160(w) x 225(d) x 85(h)
Weight (Kg)   0.65
Environmental   0 to 40°C
                     85% Relative Humidity, non condensing

Port Definitions

Line   socket for supplied lead to connect to telephone line.
Handset port   For connection of supplied handset
Headset port   4 pin socket for a headset, RJ-9

Voice Compression Card Technical Details

The VCC-8CH is a plug in card that may be installed into either the CallServer or Remote CallServer. It provides up to 8 full duplex channels of voice compression according to the G729A standard.

Dimensions (mm)   104x64x12
Weight (Kg)   0.05

"Music on Hold" Adapter Technical Details

The "Music on Hold" adapter provides a simple mechanism for injecting sound from an external device into a POTS port for use as hold music.

The adapter is 61mm x 17mm x 16mm and has an RJ45 socket and a 3.5mm stereo jack socket. The RJ45 pins 4 & 5 (i.e. centre pins) connect via a UTP cable to a POTS port. The 3.5mm socket can be connected

to the external device with an appropriate lead. Note that the left and right channels of the 3.5mm socket are shorted together.

Socket   Plug

Pin	Description
2 & 5	Shorted
3 & 4	Not connected

Due to its simplicity and the use of a POTS port there are several inherent limitations, i.e.

1. The normal POTS bandwidth filters are applied and suppress frequencies below 100Hz and above 3kHz.
2. There is no volume control in the adapter or elsewhere in the system and so this function must be supplied by the external device.
3. The POTS port presents a 600 Ohm load to the external device and requires a peak to peak audio signal of around 4V. This places some restrictions on the external device and in general the line-out port of such a device is not capable of producing the required output levels. In general the headphone or speaker outputs are more suited. An external amplifier can be used to boost the signal of an external device accordingly.

"Paging Port" Adapter Technical Details

The "Paging Port" adapter provides a simple mechanism to provide announcements on an external device connected to a POTS port.

The adapter is 61mm x 17mm x 16mm and has an RJ45 socket and a 3.5mm stereo jack socket. The RJ45 pins 4 & 5 (i.e. centre pins) connect via a UTP cable to a POTS port. The 3.5mm socket can be connected to the external device with an appropriate lead. Note that the left and right channels of the 3.5mm socket are shorted together.

Socket   Plug

Pin   Description
2 & 5   Shorted
3 & 4   Not connected

Due to its simplicity and the use of a POTS port there are several inherent limitations, i.e.
1   The normal POTS bandwidth filters are applied and suppress frequencies below 100Hz and above 3kHz.
2   There is no volume control in the adapter or elsewhere in the system and so this function must be supplied by the external device.
3   The POTS port presents a relatively high impedance source to the external device and has low output levels. This necessitates a high impedance input on the external device. It is highly recommended to test each configuration before installation to ensure appropriate output levels are attained.

Trunk NT Mode Dongles

These dongles are used to swap the transmit and receive pairs from the ISDN connector to allow a straight category 5 UTP cable to be used to connect to the slaved device, e.g. another PBX or video conferencing equipment.

The BRI dongles also provide termination resistors that are usually provided by a telephone company's NT.

The dongles may be purchased from your reseller or created as follows:

PRI Cross-over Dongle

RJ45 connector pin out	RJ45 Connector pin out
1	4
2	5
3	3
4	1
5	2
6	6
7	7
8	8

BRI Short Haul Cross-over Dongle

The short haul dongle should be used to connect to a slaved device within 10m of the trunk connector, i.e. the cable connected to the dongle should be <10m. This is to ensure signal quality. Exceeding 10m may cause a high error rate on the link and, although generally operational, is not advised.

RJ45 connector pin out	RJ45 Connector pin out
1	1
2	2
3	4
4	3
5	6
6	5
7	7
8	8

A 50 Ohm resistor is connected across pins 3 & 6 of the first RJ45 connector.
A 50 Ohm resistor is connected across pins 4 & 5 of the first RJ45 connector.

BRI Long Haul Cross-over Dongle

The long haul dongle should be used when the slaved device is further than 10m from the trunk connector. The pin out is the same as the short haul dongle but the termination resistors change, i.e.

RJ45 connector pin out	RJ45 Connector pin out
1	1
2	2
3	4
4	3
5	6
6	5
7	7
8	8

A 100 Ohm resistor is connected across pins 3 & 6 of the first RJ45 connector.
A 100 Ohm resistor is connected across pins 4 & 5 of the first RJ45 connector.

Further 100 Ohm termination is required at the slaved device to ensure signal quality. This is usually provided by either another (but different) dongle or a patch panel containing the termination. These devices are generally available from RS Components or Farnell, however for completeness the dongle can be made up as follows:

RJ45 connector pin out	RJ45 Connector pin out
1	1
2	2
3	3
4	4
5	5
6	6

7   7
8   8
(i.e. no cable swaps)

A 100 Ohm resistor is connected across pins 3 & 6 of the RJ45 connector.
A 100 Ohm resistor is connected across pins 4 & 5 of the RJ45 connector.

Glossary

Blind Transfer
This is an un-announced transfer, where the user dials the number required and hangs up to connect the caller without the destination verbally accepting the call.

Broadcast
In general, to broadcast (verb) is to cast or throw forth something in all directions at the same time. A radio or television broadcast (noun) is a program that is transmitted over airwaves for public reception by anyone with a receiver tuned to the right signal channel.

Broadcast should not be confused with a multicast which is a transmission to multiple specific receivers (as in e-mail to a distribution list).

Call Waiting
The Call Waiting facility enables a User to receive a second call and handle the two calls simultaneously.

Caller Line Identification (CLI/CLID)
CLI is a service supplied by most phone providers, whereby your phone number is transmitted when you make a phone call.

CLI codes
- 1471 - Check the number of the last person to ring you. Add a "3" to call the last number
- 141 - Dial before a phone number to withhold your number
- 1470 - Dial before a number to allow your number to be sent even if your network normally withholds your number

DHCP
Dynamic Host Configuration Protocol (DHCP) is a communications protocol that lets network administrators manage centrally and automate the assignment of IP addresses in an organisations network. Using IP, each host on the network must be assigned a unique IP address. Without DHCP, the IP address must be entered manually at each computer and, if computers move to another location in another part of the network, a new IP address must be entered. DHCP lets a network administrator supervise and distribute IP addresses from a central point and automatically sends a new IP address when a computer is connected into a different place in the network.

A DHCP client is a host on a network configured to obtain an IP address from a DHCP Server. A DHCP server is host on the network configured to assign IP addresses to DHCP clients.

DHCP uses the concept of a "lease" or amount of time that a given IP address will be valid for a computer. The lease time can vary depending on how long a user is likely to require the network connection. This is especially useful in environments where users change frequently. Using very short leases, DHCP can dynamically reconfigure networks in which there are more computers than there are available IP addresses.

DHCP supports static addresses for computers that need a permanent IP address.

Some operating systems, including Windows NT/2000, come with DHCP servers.

DDI

Direct Dial-In (DDI) in Europe and Oceania, is a feature offered by telephone companies for use with their customers' private branch exchange (PBX) systems. The telephone company provides one or more trunk lines to the customer for connection to the customer's PBX and allocates a range of telephone numbers to this line (or group of lines) and forwards all calls to such numbers via the trunk. As calls are presented to the PBX, the dialed destination number (DNIS) is transmitted, usually partially (e.g., last four digits), so that the PBX can route the call directly to the desired telephone extension within the organization without the need for an operator or attendant. The service allows direct inward call routing to each extension while maintaining only a limited number of subscriber lines to satisfy the average concurrent usage of the customer.

DDI trunks must be powered by the customer premises equipment. The central office equipment detects the power state of the line and disables service if the circuit is not powered up. This is the reverse arrangement from standard plain old telephone service (POTS) lines which are powered by the central office.

The corresponding service to DDI for outgoing calls from a PBX to the central office exchange is called *direct outward dialing* (DOD) or *Direct Dial Central Office* (DDCO). This service is often combined with DDI service and allows direct dialing of global telephone numbers by every extension covered by the service without the assistance of an operator. The caller line identification (CLI) or *caller-id* of extensions for outgoing calls is often set to the extension DDI number, but may be the organization's central switch board number.

Domain Characters

Alpha-numeric characters only, no punctuation, beginning with a letter

DPNSS

The Digital Private Network Signalling System (DPNSS) has evolved as the de-facto standard for inter-site communications between traditional PBXs in the UK. Whilst not formally regulated, it is a voluntary standard that has been developed by BT in conjunction with major PBX manufacturers over the last two decades and allows interconnection of equipment from different vendors over the ISDN network.

Gatekeeper

A gatekeeper controls the access into and out of the network ensuring the information is correctly formatted, translating one transmission type into another

Gateway

A gateway is a network point that acts as an entrance to another network. The computers that control traffic within a network are gateway nodes. In the network a computer server acting as a gateway node is often also acting as a proxy server and a firewall server. A gateway is often associated with both a router, which knows where to direct a given packet of data that arrives at the gateway, and a switch, which furnishes the actual path in and out of the gateway for a given packet.

Global User

A User not assigned to a specific Company and is able to view all Users and Departments via their PCS directories.

GUID (global unique identifier)

A unique string of digits generated to identify an entry within the database such as a User, Department etc, eg eb3c0700-3386-e813-8567-0007d9001823

H.323

H.323 is a standard approved by the International Telecommunication Union (ITU) in 1996 to promote compatibility in videoconference transmissions over IP networks. H.323 was originally promoted as a way to provide consistency in audio, video and data packet transmissions in the event that a local area network (LAN) did not provide guaranteed service quality (QoS). It is now considered to be the standard for interoperability in audio, video and data transmissions as well as Internet phone and voice-over-IP (VoIP) because it addresses call control and management for both point-to-point and multipoint conferences as well as gateway administration of media traffic, bandwidth and user participation.

Hub

In general, a hub is the central part of a wheel where the spokes come together. The term is familiar to frequent fliers who travel through airport "hubs" to make connecting flights from one point to another. In data communications, a hub is a place of convergence where data arrives from one or more directions and is forwarded out in one or more other directions. A hub usually includes a switch of some kind. (A product that is called a "switch" could usually be considered a hub as well.) The distinction seems to be that the hub is the place where data comes together and the switch is what determines how and where data is forwarded from the place where data comes together. Regarded in its switching aspects, a hub can also include a router.

LDAP

LDAP (Lightweight Directory Access Protocol) is a software protocol for enabling anyone to locate organizations, individuals, and other resources such as files and devices in a network. LDAP is a "lightweight" (smaller amount of code) version of Directory Access Protocol (DAP), which is part of X.500, a standard for directory services in a network. LDAP is lighter because in its initial version it did not include security features. LDAP originated at the University of Michigan and has been endorsed by at least 40 companies. Netscape includes it in its latest Communicator suite of products. Microsoft includes it as part of what it calls Active Directory in a number of products including Outlook Express. Novell's NetWare Directory Services interoperates with LDAP.

In a network, a directory tells you where in the network something is located. On TCP/IP networks (including the Internet), the domain name system (DNS) is the directory system used to relate the domain name to a specific network address (a unique location on the network). However, you may not know the domain name. LDAP allows you to search for an individual without knowing where they're located (although additional information will help with the search).

An LDAP directory is organized in a simple "tree" hierarchy consisting of the following levels:

    The root directory (the starting place or the source of the tree), which branches out to
    Countries, each of which branches out to
    Organizations, which branch out to
    Organizational units (divisions, departments, and so forth), which branches out to (includes an entry for)
    Individuals (which includes people, files, and shared resources such as printers)

An LDAP directory can be distributed among many servers. Each server can have a replicated version of the total directory that is synchronized periodically. An LDAP server is called a Directory System Agent (DSA). An LDAP server that receives a request from a user takes responsibility for the request, passing it to other DSAs as necessary, but ensuring a single coordinated response for the user.

Multicast

Multicast is communication between a single sender and multiple specific receivers on a network (as in e-mail to a distribution list).

No Answer Time
The amount of time an extension will ring before it is considered not answered. When configured for a User the call will route to either the Forward on No Answer number or voicemail. If configured for a Department the call will route to the next member in the Group.

Power over Ethernet
Power over Ethernet (PoE) is a technology for wired Ethernet LANs (Local Area Networks) that allows the electrical current, necessary for the operation of each device, to be carried by the data cables rather than by power cords. This minimizes the number of mains connections required to install the network. The result is lower cost, less downtime, easier maintenance, and greater installation flexibility than with traditional wiring. The alternative is to power each device directly from individual power supplies. Power over Ethernet will power devices up to 100 meters from the power supply and the number of devices supported by the PoE supply is a factor of the devices power requirement.

Quality of Service
On the Internet and in other networks, QoS (Quality of Service) is the idea that transmission rates, error rates, and other characteristics can be measured, improved, and, to some extent, guaranteed in advance. QoS is of particular concern for the continuous transmission of high-bandwidth video and multimedia information. Transmitting this kind of content dependably is difficult in public networks using ordinary "best effort" protocols.

Using the Internet's Resource Reservation Protocol (RSVP), packets passing through a gateway host can be expedited based on policy and reservation criteria arranged in advance. Using ATM, which also lets a company or user pre-select a level of quality in terms of service, QoS can be measured and guaranteed in terms of the average delay at a gateway, the variation in delay in a group of cells (cells are 53-byte transmission units), cell losses, and the transmission error rate.

The Common Open Policy Service (COPS) is a relatively new protocol that allows router and layer 3 switches to get QoS policy information from the network policy server.

Session Initiation Protocol (SIP) 2.0
Session Initiation Protocol (SIP) is a protocol for initiating, modifying, and terminating an interactive user session that involves multimedia elements in this instance Voice over IP. SIP is a peer-to-peer protocol and as such it requires only a very simple (and thus highly scalable) core network with intelligence distributed to the network edge, embedded in endpoints (terminating devices such as IP Phones built in either hardware or software). Many SIP features are implemented in the communicating endpoints as opposed to traditional telephony features, which are commonly implemented in the network or PBX. A direct alternative to H.323, SIP has its roots in the IP community rather than the telecom industry. SIP has been standardised and governed primarily by the IETF while the H.323 VoIP protocol has been traditionally more associated with the ITU. Like H.323, SIP works in concert with several other protocols and is only involved in the signalling portion of a communication session. SIP acts as a carrier for the Session Description Protocol (SDP), which describes the media content of the session, e.g. what IP ports to use, the codec being used etc. In typical use, SIP "sessions" are simply packet streams of the Real-Time Transport Protocol (RTP). RTP is the carrier for the actual voice or video content itself and is also used by H.323. SIP is similar to HTTP and shares some of its design principles: It is human readable and request response structured. SIP shares many HTTP status codes, such as the familiar '404 not found'. SIP and H.323 are not limited to voice communication but can mediate any kind of communication session including video, instant messaging, online games, and virtual reality, or future, unrealised applications. This fully integrated feature within maximiser allows SIP and H.323 devices to be concurrently supported across a single system, with calls being passed seamlessly between both device types. The following

features are supported for SIP end point devices; Simple Call Transfer - Refer Method (RFC 3515), Message Waiting Indication (RFC 3842), Session Timers (RFC 4028), RTP Payload for DTMF Digits (RFC 2833).

Switch

In telecommunications, a switch is a network device that selects a path or circuit for sending a unit of data to its next destination. A switch may also include the function of the router, a device or program that can determine the route and specifically what adjacent network point the data should be sent to. In general, a switch is a simpler and faster mechanism than a router, which requires knowledge about the network and how to determine the route.

Index

SpliceCom

Defining next-generation communication