SIP Communicator Administrator Guide

Version 5

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Contents

1. Logging on as Administrator:

Browse to: https://pbxX.spitfiretsp.net/ *Where X = PBX number where domain is configured*

Enter the admin user account: 0000@subdomain.spitfiretsp.net
Enter the admin password: Supplied by Spitfire

2. Changing the Administrator password:

- Select Accounts from the menu bar:
- Click Edit next to the Admin account.
- Change the Web Password.
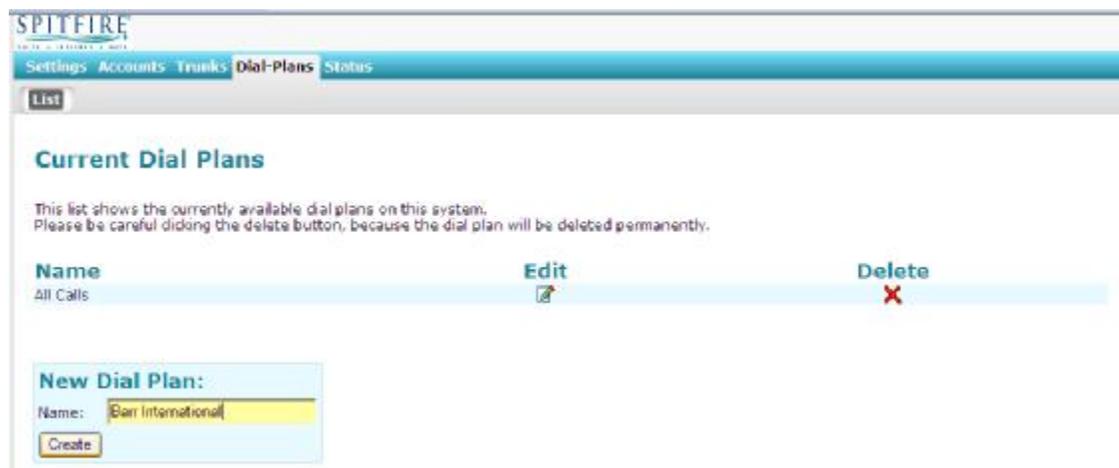- Click Save.

3. Domain Settings:

- Used to set default settings and timers for the domain.

- Voicemail Timeout: Number of seconds that a call will ring at an extension before going to voicemail. *Users can over-ride this setting.*
- Voicemail Size: Number of messages that can be stored.
- Mailbox Escape Account: Where calls should be directed if caller dials "0" while listening to mailbox greeting. *Users can over-ride this setting*
- Mailbox Direct Dial Prefix: Dial this digit in front of the extension number to call/transfer a call directly to the mailbox.
- Mailbox Explanation Prompt: Turns system greeting on or off.
- Call Forward On No Answer Timeout: Default time calls rings at extension before being diverted if call-forward no-answer is set. *Users can over-ride this setting.*
- Park Reminder: Will ring extension if parked call is forgotten.
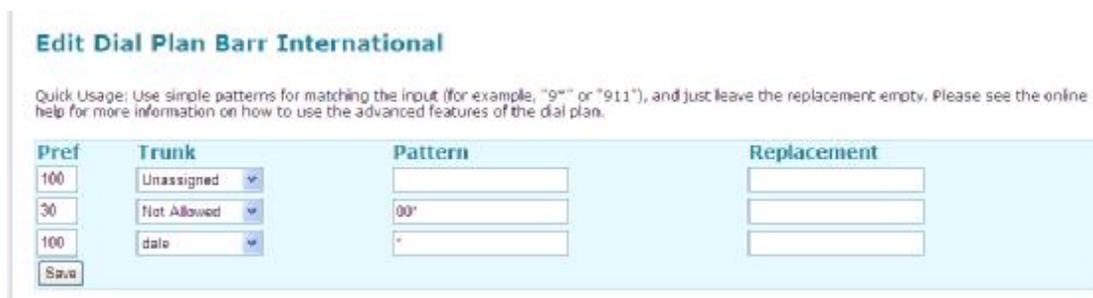
4. Creating Dial Plans

- Select Dial-Plans from the menu bar:
- Type in the name for the Dial Plan and click Create as shown below.

- Click Edit for the Dial Plan just created.
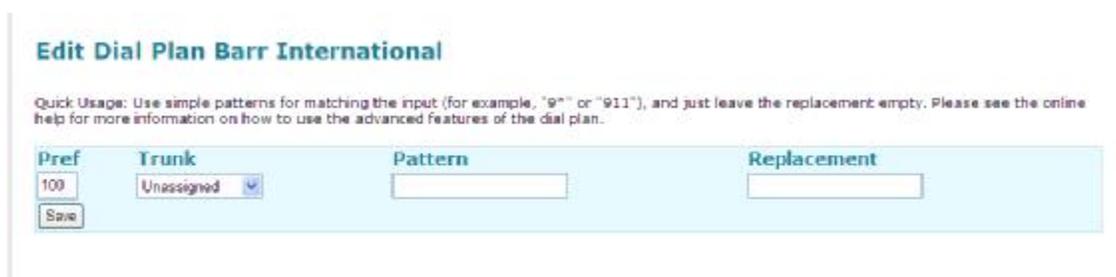
- Overwrite default entries with preferred dialling patterns and click "Save"

Note: In this example all international calls are barred.

- Pref: The order in which applied rules are checked. Rules with lower Pref are checked 1st.
- Trunk: Select you domain trunk or other routing options from the drop down list.
- Pattern: Dialling pattern checked by system. (* indicates all numbers)
- Replacement: Replacement dial string.

5. Creating Extensions:

- Select Accounts from the menu bar.
- Select Create.

- Select Extension under Account Type.
- Select a Dial Plan to be used by the extension/s.
- Enter in the details for up to 10 extensions simultaneously.
- Click Create.
- Number: Enter the extension number and DDI (If assigned) separated by a "/" *Note: when creating an extension separating the extension and DDI numbers with a "/" will create 1 account with two aliases using a space will create two separate accounts. The "/" should only be used when creating extensions, if adding a DDI to an extension when editing it, then use a space. Do not add the leading zero of the DDI. After creating return to "List" and ensure that you see the two entries for the extension created: 2003   Extension(Alison Clark)*
  2075087862 <2003>   Extension(Alison Clark)
- Name: This is the name displayed when you call another user on the domain. The name displayed on the handset in "idle" state is configured on the handset.
- SIP Pass: Used by handset to register.
- Web Pass: Used to access user web interface.
- Pin: Used to access voicemail remotely.
- Email: For voicemail to email and other notifications.
- To edit an extension select Lists under Accounts.
- Select Edit.

Editing Extension 2005 207141996

Account Number(s):	2005 207141996
Dial Plan:	Barr international
ANI:	
Send daily CDR report to:	
Show following ACD queues:	
First name (e.g. John):	Alison
Last name (e.g. Smith):	Clarke
SIP Password:	••••••••••
SIP Password (repeat):	••••••••••
Web Password:	••••••••••
Web Password (repeat):	••••••••••
PIN (e.g. 1234):	••••••
PIN (repeat):	••••••
Timezone:	Default Time Zone
IVR Language:	Default System IVR Language
Web Language:	Default System Web Language
Upload picture:	Browse...
Wakeup time (e.g. 7:15):	
Watch the calls of the following extensions (* for all):	
Watch the presence of the following extensions (* for all):	
Watch following accounts on PAC:	
Explicitly specify park orbit preference:	
Explicitly specify pickup preference:	
Permissions to monitor this account:	
Block outgoing caller-ID:	⊙ no ○ yes

- ANI: Outgoing CLI
- Upload Picture: Picture displayed on users homepage.
- Explicitly specify park orbit: Can specify park location (must be an extension)
- Block outgoing caller-ID: Withholds extension CLI if set to "Yes".

- Redirection, Mailbox and Email are covered in detail in the Spitfire Web User Guide.
- Registration: Displays status of handset registration.
- Permission: Allows the extension to be set as an Administrator and over-ride DND set on another extension.

6. Configuring Handsets

Note: If the handsets are supplied by Spitfire then they will automatically upgrade to the latest firmware version and download a Spitfire Default Configuration when connected to a network offering DHCP and with internet access. The only configuration required to get a handset working is the Identity settings and Handset buttons.

Remote access to handsets:

The routers used for SIP Communicator are configured by default to use the DHCP on the 10.0.0.0 to network. There is also port forwarding set up as follows:

Port	Destination IP
8001	10.0.0.1
8002	10.0.0.2
8003	10.0.0.3

……… and so on

Accessing a handset remotely

- Ask the user to press the "?" button on the handset to see the IP address.
- Enter the router address and port number for the handset IP address into a browser.
  Example: http://86.223.17.25:8001/
- You will then be directed to the login page of the handset on IP address 10.0.0.1.

Configure a SIP Account on a SNOM handset:

- Once handset has booted, press the "?" button to view the IP address.
- Type the IP address into a browser and enter the user name and password to log onto the handset interface. (User name and password set by Administrator)
- Select "Identity 1" and configure the SIP Registration details.
  - Display Name: Name Displayed on handset display. (recommend only 1st name for 320 and 300)
  - Account: Extension number as configured on the SIP Communicator.
  - Password: SIP Password as set on the SIP Communicator.
  - Registrar: Company Domain Name.
  - Outbound Proxy: pbx2.spitfiretsp.net.
  - Authentication Username: Same as Account
  - Display text for idle screen: Overwrites Display Name.

- Click "Save"; the handset should resister and be operational.
- Click "System Information" and ensure that the account has registered.

Programming Buttons on a Snom:

- Select Function Keys

Service Flags:	Event = Extension:	Dial = Service Flag Extension Number
Record Shared Mailbox Greeting:	Event = Extension:	Dial = Mailbox Extension Number
Record AA Greeting:	Event = Speed Dial	Dial = *98 + AA Extension Number
Pick Up:	Event = Speed Dial	Dial = *87
Extension	Event = Speed Dial	Dial = Extension Number
Call Park	Event = Speed Dial	Dial = *85
Park Retrieve	Event = Speed Dial	Dial = *86

Configure a SIP Account on a Yealink handset:

- Once handset has booted, press the "OK" button to view the IP address.
- Type the IP address into a browser and enter the user name and password to log onto the handset interface. (User name and password set by Administrator)
- Select "Account"

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- Account Active: Select "On"
- Label: Display name on handset (For a T28 set this to the extension number)
- Display Name: Name displayed when you call another user. (This is overwritten by the PBX)
- Register Name: Extension Number
- User Name: Extension Number
- Password: SIP password as set on the PBX.
- SIP Server: Domain where account is configured.
- Enable Outbound Proxy Server: Enable
- Outbound Proxy: PBX where domain is configured.
- Voicemail: Extension Number
- Click "Confirm"
- Ensure that the "Register Status" is "Registered"

Programming Buttons on a Yealink:

- Select "Phone"
- Select "DSS Key"

Note: Memory Keys are the 10 buttons down the side of the phone on a T28 and T26 which are configurable. The Line keys are the buttons next to the display and it is recommended that these are not changed from the default set by the configuration file installed at boot up.

Memory Key >>

Key	Type	Mode	Line	Extension	Pickup Number
DSS Key 1	BLF	Conference	Line 1	1	
DSS Key 2	BLF	Conference	Line 1	2	
DSS Key 3	BLF	Conference	Line 1	200	
DSS Key 4	BLF	Conference	Line 1	2002	*872002
DSS Key 5	N/A	Conference	Line 1		
DSS Key 6	KeyEvent	Forward	Line 1		
DSS Key 7	Speed Dial	Conference	Line 1	*85	
DSS Key 8	N/A	Conference	Line 1		
DSS Key 9	Speed Dial	Conference	Line 1	*87	
DSS Key 10	Speed Dial	Conference	Line 1	*85	

Line Key >>

	Type	Line	Extension	Pickup Number
Service Flags:	BLF	1	Ext No of SF	
Record Shared Mailbox Greeting:	BLF	1	Mailbox Ext No	
Record AA Greeting:	Speed Dial	1	*98 + AA Extension Number	
Pick Up:	Speed Dial	1	*87	
Extension	Speed Dial	1	Extension Number	*87+Extension No
Call Park	Speed Dial	1	*85	
Park Retrieve	Speed Dial	1	*86	

7. Configuring Service Flags:

- Select Create under Accounts.
- Select Service Flag from the drop-down menu.
- Enter an extension number.
- Click Create.
- Select List under Accounts.
- Select Edit next to the service flag.

Edit Service Flag Time1

Please specify the service times in the format HH:MM-HH:MM, you may use the symbol "P" for PM or use the 24-hour format. You may specify any number of intervals for each day (for example, "9:00-12:30 1:00P-5:00P").
Holidays are written in the format Month/Day, you may also use any number of holidays (for example, "12/24 12/25 12/26").

Identity:

Account Number(s): 9010

Description:

Mode: Manual

Current State: Clear

Display Name: Day Service

Extensions that may change status: 2031 2002 2005

Permissions to monitor this account:

[Save]

- Mode: Manual or Day/Night. (see notes below)
- Current State: Activate/Deactivate.
- Extensions that may change status: User with permission to activate/deactivate flag.

Day/Night Mode: System automatically changes status of flag at predefined times of the day.

Edit Service Flag 9000

Please specify the service times in the format HH:MM-HH:MM, you may use the symbol "P" for PM or use the 24-hour format. You may specify an number of intervals for each day (for example, "9:00-12:30 1:00P-5:00P").
Holidays are written in the format Month/Day, you may also use any number of holidays (for example, "12/24 12/25 12/26").

Identity:

Account Number(s): 9000

Description:

Mode: Day/Night

Display Name: Night Mode

Permissions to monitor this account:

Monday: 9:00-6:00P

Tuesday: 9:00-6:00P

Wednesday: 9:00-6:00P

Thursday: 9:00-6:00P

Friday: 9:00-6:00P

Saturday:

Sunday:

Holidays: 12/25 12/26 01/01

[Save]

- Enter in the times for each day, when the flag is "clear" i.e. not set.
- Enter in any holidays where the night service setting should be used.

In this example any Hunt Group or AA with this flag assigned would route calls normally between the hours of 9am and 6pm. All calls routed to the Group or AA out of these hours will be directed to the night service destination set on the particular group or AA.

8. Configuring Group Mailboxes:

- Create an extension as described in section 3. (Call it Night Mailbox for example)
- Select List under Accounts and then select Edit for the extension created.
- Select Redirection and ensure that Do not disturb is set to "On" and click Save.

- Add the account to "Identity 2" of all handsets that should be notified of messages.
- Configure mailbox settings as described in the Spitfire Web User Guide.
- To record the greeting for the mailbox, dial the extension number, then press * and enter the PIN number. Follow prompts to record greeting.

9. Configuring Auto Attendants:

- Select Create under Accounts.
- Select Auto Attendant from the drop-down menu.
- Enter an extension number.
- Click Create.

- Select List under Accounts.
- Select Edit next to the Auto Attendant.

Edit Attendant 2201 2075015678

- Account Number/s: Enter in any DDI number associated to the AA. (separated by space)
- Name: Name of the AA.
- Extension Input: Allows callers to dial an extension directly if they know the number.
- Accounts that cannot be called: Prevents extension been called directly.
- Accounts that may record a message": Gives extensions permission to record the AA greeting.
- Dial Plan for outbound calls: Dial plan to use if destination is an off switch number.
- ANI: Outbound CLI.
- Timeout handling: This field determine how a call is handled if the caller does not make a selection and how long the system will wait for a caller to make a selection.
- Night Service: Assigns a Service Flag to the AA and the destination where calls should be routed if the flag is active.
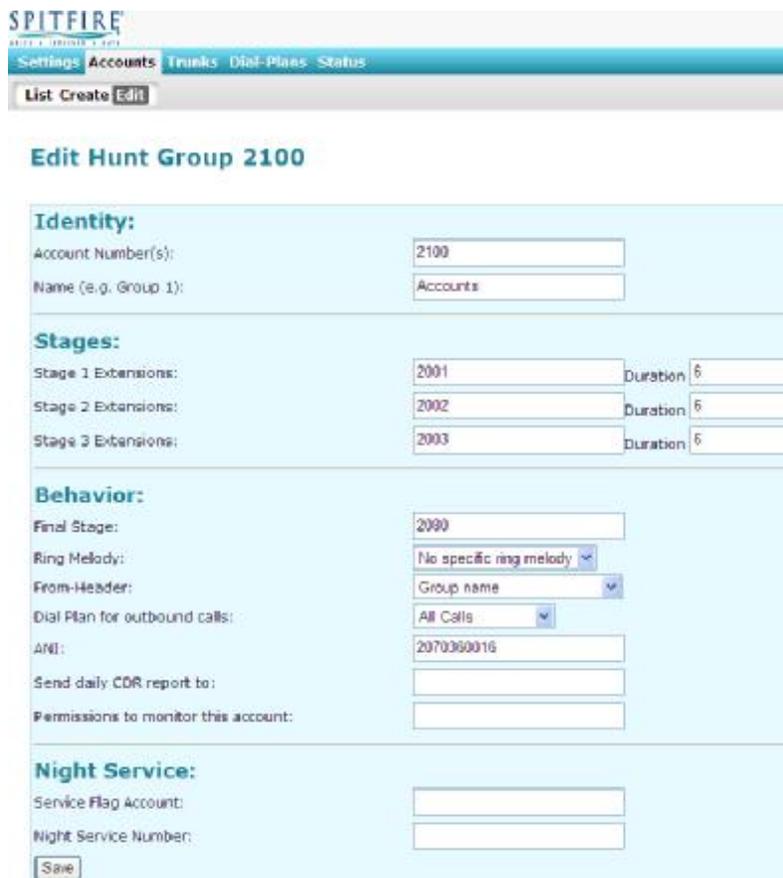- Direct Destinations: Specifies inputs and destinations for the AA.

10. Configuring Hunt Groups

- Select Create under Accounts.
- Select Hunt Group from the drop-down menu.
- Enter an extension number.
- Click Create.
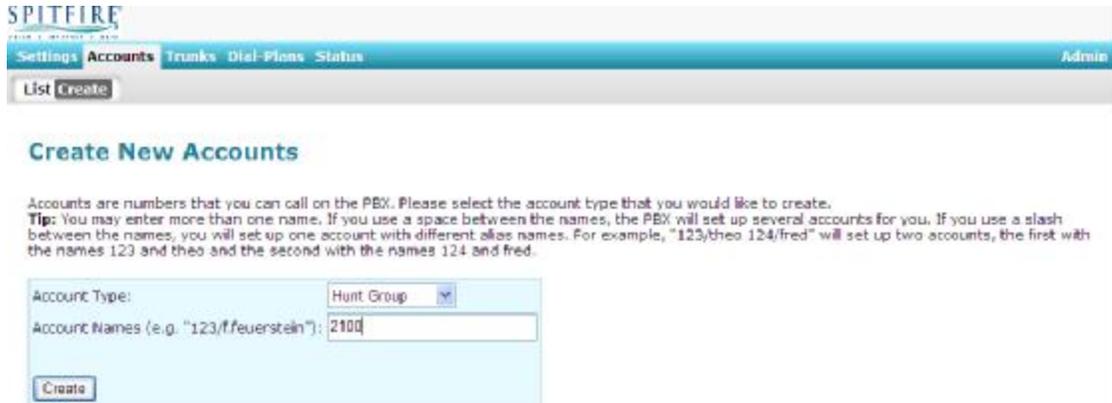
- Select List under Accounts.
- Select Edit next to the Hunt Group.

- Account Number/s: Enter in any DDI number associated to the AA. (separated by space)
- Name: Name of the AA.
- Stages: Enter extensions that should ring at each stage (separated by space) and the time for the stage ring.
- Final Stage: Enter destination if call is not answered by any extensions in the 3 stages.
- From Header: Determines what is displayed on the handset when the hunt group is called.
- ANI: Outbound CLI.
- Dial Plan for outbound calls: Dial plan to use if destination is an off switch number.
- Night Service: Assigns a Service Flag to the Hunt Group and the destination where calls should be routed if the flag is active.

11. Configuring Conference Bridges

- Select Create under Accounts.
- Select Conference from the drop-down menu.
- Enter an extension number.
- Click Create.

- Select List under Accounts.
- Select Edit next to the Conference Bridge.

- Account Number/s: Enter in any DDI number associated to the Conference Bridge. (separated by space)
- Name: Name of the Conference Bridge.
- Mode: Select "Ad Hoc".
- Before entering the conference: Callers can be requested to record their names 1st which will then be played to other callers already in the conference.
- Participant Access Code: Password that callers should use to enter the Conference.

12. Configuring Calling Cards:

- Select "Accounts/Create/Calling Card". Assign a number to the calling card and click "Save".

- Return to "List" and click the edit button next to the newly created calling card to display the screen show in figure 2. Fill in the details and click "Save"

- Account Number(s): Extension number for the calling card. A DDI can also be assigned to dial the calling card directly from external devices.
- Dial Plan: Assign a dial plan for the calling card to use.
- ANI: CLI sent when an external call is made from the calling card.
- Send daily CDR report to: PBX will send an email on a daily basis listing all calls made using the calling card.
- Callback: Set to "On" if calling card will be used for call backs, set to "Off" if it will be used as a standard calling card.
- Allow calling accounts on PBX: Leave set to "Block"
- Entering telephone numbers: Leave set to "User must press pound"
- Permissions to monitor account: If left blank then all users can monitor the calling card use via a button on the handset, can restrict monitoring to certain extensions by inserting extension numbers.
- All other fields are not relevant and should be left as default.

13. Assigning Barge-In Permissions:

- Select "Accounts" and then edit the extension you want to assign barge in permissions to. Then click on the permissions tab, as show in figure 1.

Figure 1

- Enter in the extension number/s (separated by a space) that this user may barge into. To barge into all extension enter **
- Click Save

Installing the TAPI Driver:

- The TAPI driver can be installed from the following link:
  http://www.pbxnsip.com/download-software/software.php
- Click on "Run" and install the Driver using the install wizard.
- Reboot the PC. **(Not prompted for a reboot but is required)**

Configuring the TAPI Driver:

- Open Outlook and select Contacts:
- Click on "Auto Dialer" as shown in figure 1.
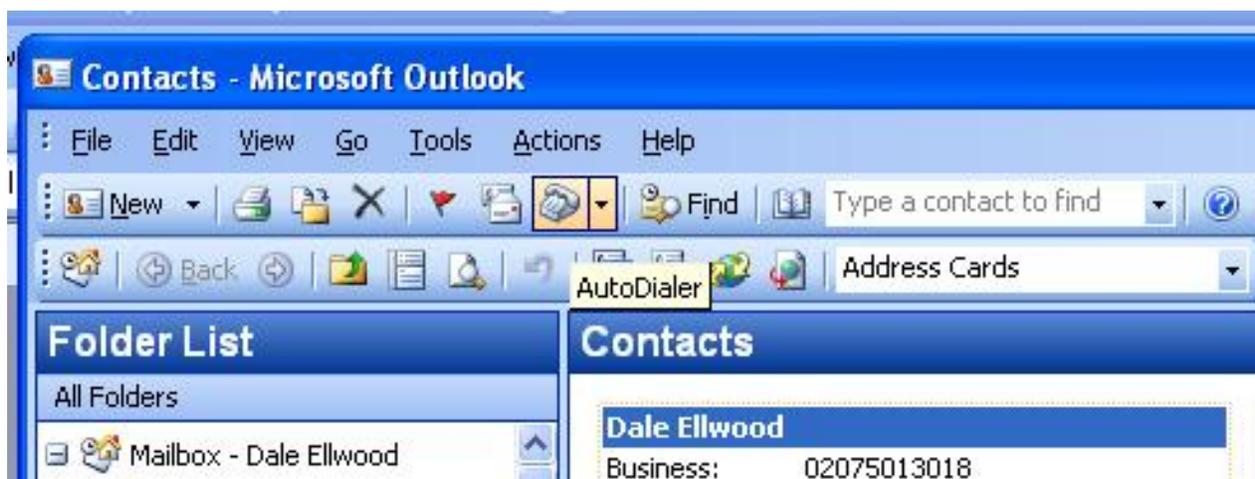
Figure 1

- On the "New Call" screen, click "Dialing Options"
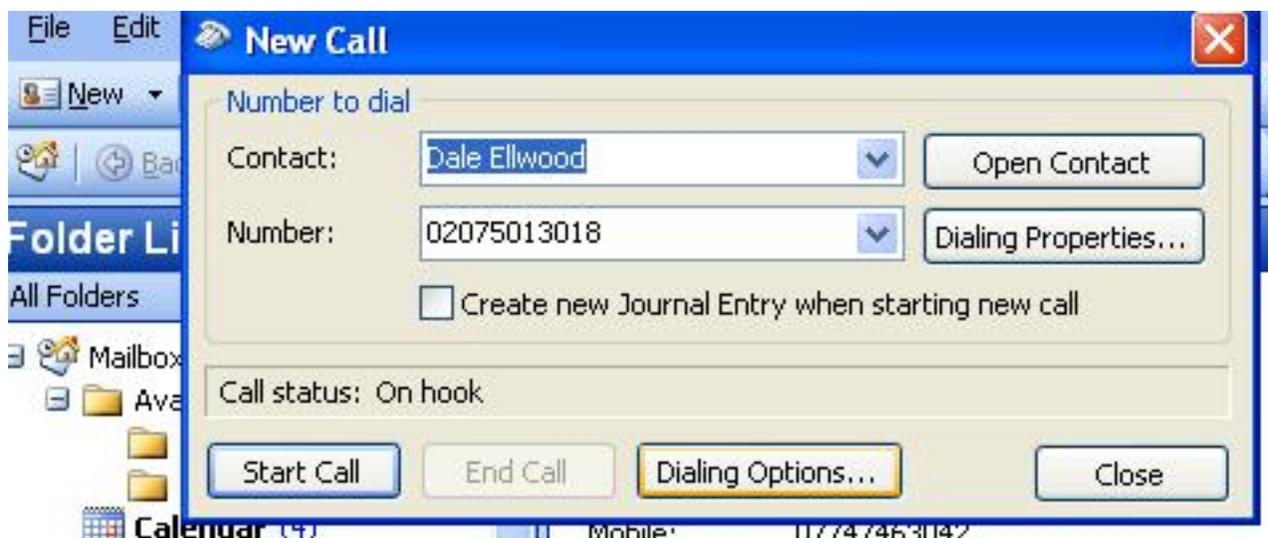
Figure 2

- On the Dialing Options page, select "pbxnsip PBX" from the drop down list under the "Connect using line" as shown in Figure 3.

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- Click on "Line Properties"
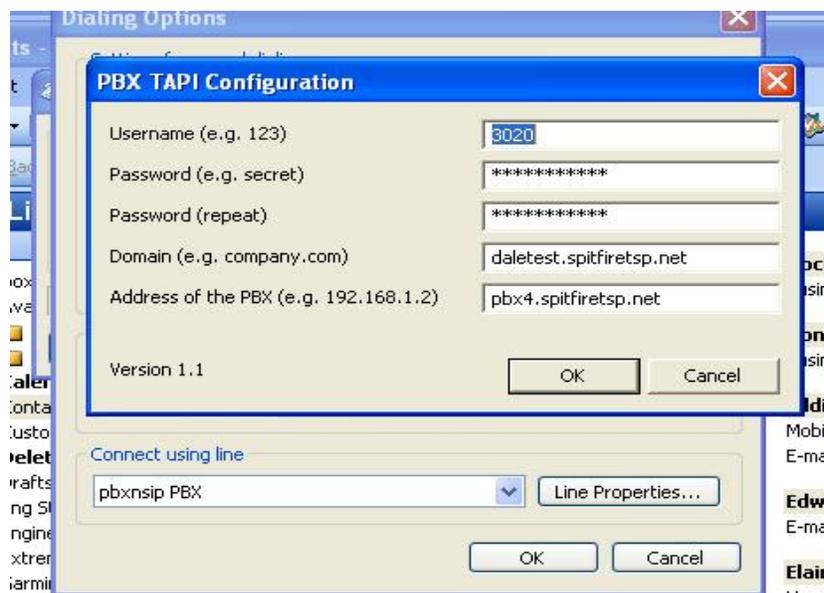
- Fill in the SIP account details and Click "OK"
  - Username: Extension Number
  - Password: SIP Account password; can be obtained from the system administrator.
  - Domain: Company Domain where SIP account is configured.
  - Address of the PBX: PBX where Company domain is configured.
- Click OK and close the "New Call" screen.

Dialing from Outlook Contacts:

- Select the contact in Outlook.
- Click "Auto Dialer"
- Click "Start Call" as shown in figure 5.
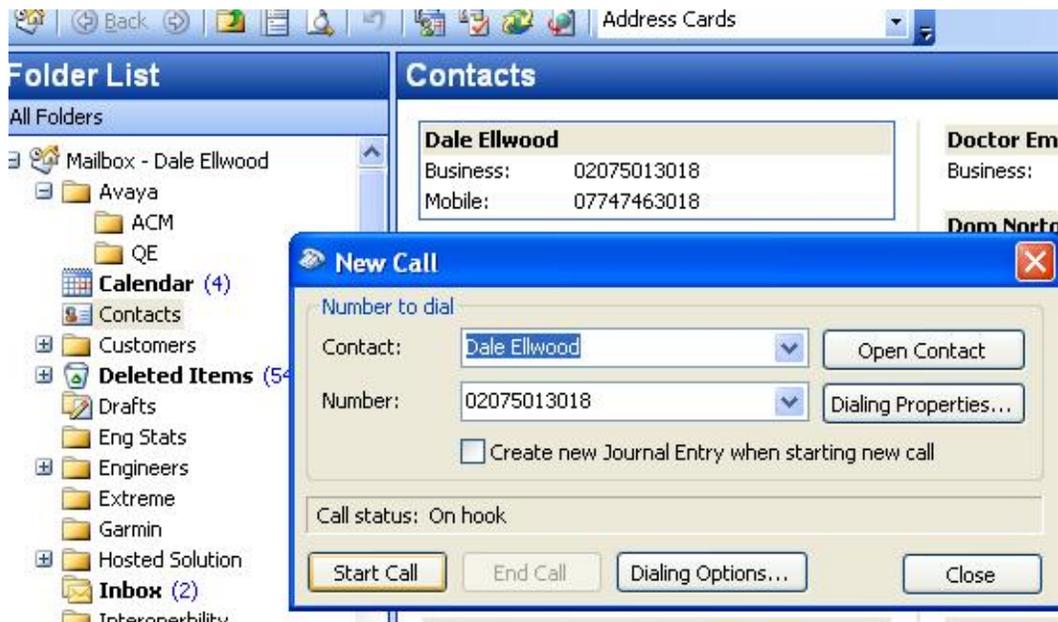
Figure 5

- Your handset will ring, pick up the receiver or press the speaker button to connect the call.
- Close the "New Call" screen.

14. Programming Tips:

- Consider the numbering plan carefully and consider future expansion.
- If possible match digits of the extension number to the last few digits of the DDI.
- Use the same number scheme for different types of accounts:
  Example:
  Extensions: 2000 to 2099
  Hunt groups: 2100 to 2199
  Auto Attendants: 2200 to 2299
  Group Mailboxes: 2300 to 2399
- Create Dial Plans 1st.
- Create extensions and configure handsets, once all handsets "up and running" start with system configuration.
- "Work Backwards" Example: When setting up AA or HG, create all destinations, service flags and group mailboxes needed for the AA or HG 1st. This will prevent you "jumping" around the configuration.

Programming Tips continued……….

- Always set a final destination in Hunt Groups, even if the customer is adamant that someone will answer the call.
- When setting up an AA using the "Dial extension" feature ensure the options you configure in the AA do not clash with the extension numbers.

13. Glossary

|	
Auto Attendant	An automated system designed to guide a caller through the options of a voice menu. Typically set to answer and route incoming calls.
Blind Transfer	Where one user transfers a call to another user without notifying the recipient. It is also known as unsupervised transfer or cold transfer.
Call Forward	A feature that allows an incoming call to a called party which would be otherwise unavailable, to be redirected to a mobile telephone or other telephone number where the desired called party is situated.
Call Hold	Allows a system user to manually place a call on hold, so the incoming caller cannot hear your conversation.
Call Pick Up	Call Pickup allows a user to answer a call presented to another extension.
Call Transfer	Call Transfer allows another party to be placed on hold and transferred to another destination number.
Call Waiting	A feature that allows a user to be notified of another incoming call while a call is already in progress, and gives the user the ability to answer the other call.
Codec	Codec is a term that arises from the Compression -Decompression or enCOder/DECoder process. It is used for software or hardware devices that can convert or transform a data stream. An audio codec converts analog audio signals into digital signals for transmission or storage. A receiving device then converts the digital signals back to analog using an audio decompressor, for playback.
Conference Bridge	Allows both internal and external telephone users to dial into a pre-configured conference set up by the system administrator.
DDI	Direct Dialing Inward. It allows a caller outside a company to call an internal extension, or group of extensions, without going through the switchboard.
Domain	A company's individual partition on the hosted telephone system.
Do Not Disturb	This is the ability to temporarily stop incoming calls to a user's telephone.

DSL	Digital Subscriber Line. A family of digital telecommunications protocols designed to allow high speed data communication over the existing copper telephone lines between end-users and telephone companies. Also known as xDSL.
G.711	3 kHz audio encoded at 64-kbps. G.711 is PCM audio, the format used for voice delivery over traditional telephone networks and exchanges.
Hot Desking	Enables a telephone user to log in and out of any telephone on the system.
Hunt groups	A Hunt Group is a collection of users handling similar types of calls, e.g. a sales department. An incoming caller wishing to speak to Sales can ring one number but the call can be answered by any number of extensions that are members of the Hunt Group.
Jitter	A term used to describe the variance in delay/latency of transmitted data packets.
Lag	Lag is the term used to indicate the extra time taken by a packet of data to travel from the source computer to the destination computer and back again. The lag may be caused by poor networking or by inefficient or excessive processing.
Latency	Latency the time taken for a packet of data to be sent by an application, travel and be received by another application.
Night Service	The period outside normal working hours when the call centre is either directing or overflowing calls to another answering point.
Overflow Group	If all extensions in the Hunt Group are busy or not answered, another Hunt Group, called an Overflow Group, can be used to take the calls. An overflow time can be set to stipulate how long a call will queue before being passed to the Overflow Group.
Packet	A logically grouped unit of data. Packets contain a payload (the information to be transmitted), originator, destination and synchronizing information. The idea with packets is to transmit them over a network so each individual packet can be sent along the most optimal route to its. Packets are assembled on one end of the communication and re-assembled on the receiving end based on the header addressing information at the front of each packet. Routers in the network will store and forward packets based on network delays, errors and re-transmittal requests from the receiving end.
Packet loss	Packet loss is the term used to indicate the loss of data packets during transmission over a computer network. This may happen on account of high network latency or on account of overloading of switches or routers that is unable to process or route all the incoming data.
Packet Switching	A means of economically sending and receiving data over alternate, multiple network channels. The premise for packet switching is the packet, a small bundle of information containing the payload and routing information. Packet switching takes data, breaks it down into packets, transmits the packets and does the reverse on the other end. Packets can be sent in order and then be received in a different order - only to be put back in the correct order in seconds. Although traditionally used for data, packet networks, especially well-managed ones, are becoming suitable for real-time transmission of voice and video.
PBX	Private Branch exchange. A telephony/data switching system, usually located on customer premises and belonging to the user. In contrast to Centrex which is largely located on the common carrier's premises?
PoE	Power Over Ethernet (PoE) describes a system to transmit electrical power, along with data, to remote devices over standard twisted-pair cable in an Ethernet network.
PSTN	Public Switched Telephone Network. The combination of local, long-distance, and international carriers that make up the worldwide telephone network
QoS	Quality of Service - the ability of a network (including applications, hosts, and infrastructure devices) to deliver traffic with minimum delay and maximum availability.
Redial	A service feature that allows the user to dial, by depressing a single key or a few keys, the most recent telephone number dialed.
RTP	Real Time Protocol (RTP) is a packet based communication protocol that adds timing and sequence information to each packet to allow the reassembly of packets to reproduce real time audio and video information. RTP is a transport used in IP audio and video environments.
SIP	Session Initiation Protocol. An emerging Internet standard to facilitate mixed-media sessions over IP. Sessions may span the range of services from voice-only to full multimedia conferencing; multipoint is also included.
Soft phone	IP telephony software that lets users send and receive calls from non-dedicated hardware, such as a PC. It is typically used with a headset and microphone.

TDM	Time-division multiplexing (TDM) is a method of putting multiple data streams in a single signal by separating the signal into many segments, each having a very short duration. Each individual data stream is reassembled at the receiving end based on the timing.
Twinning	Allows an external telephone such as mobile phone/home phone or soft phone to ring simultaneously with calls that come into your office phone
VoIP	The process of making and receiving voice transmissions over any IP network. IP networks include the Internet, office LANs, and private data networks between corporate offices. The main advantage of VoIP is that users can connect from anywhere and make phone calls without incurring typical analog telephone charges, such as for long-distance calls.
WES	Wholesale Extension Service (WES) is a fully managed fibre connection presented on Ethernet providing a high-speed, permanent, point-to-point circuit, offering secure and reliable bandwidth for the transmission of data, voice and video. This is typically available in 10Mb, 100Mb or 1Gb versions.