

SIP Communicator Administrator Guide



Version 5

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1. Logging on as Administrator:

Browse to: <https://pbxX.spitfiresp.net/> Where X = PBX number where domain is configured

A screenshot of the PBX Login page. At the top, there is a key icon and the text: "Please enter your login information. For more information about the login procedure, please see the [online documentation](#)." Below this is the "PBX Login" form. It contains four input fields: "Account:" (text), "Password:" (text), "Login Type:" (dropdown menu with "Automatic" selected), and "Language:" (dropdown menu with "English" selected). There is a "Login" button below the fields. At the bottom left, there is a checkbox labeled "Remember login information."

Enter the admin user account: 0000@subdomain.spitfiresp.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:

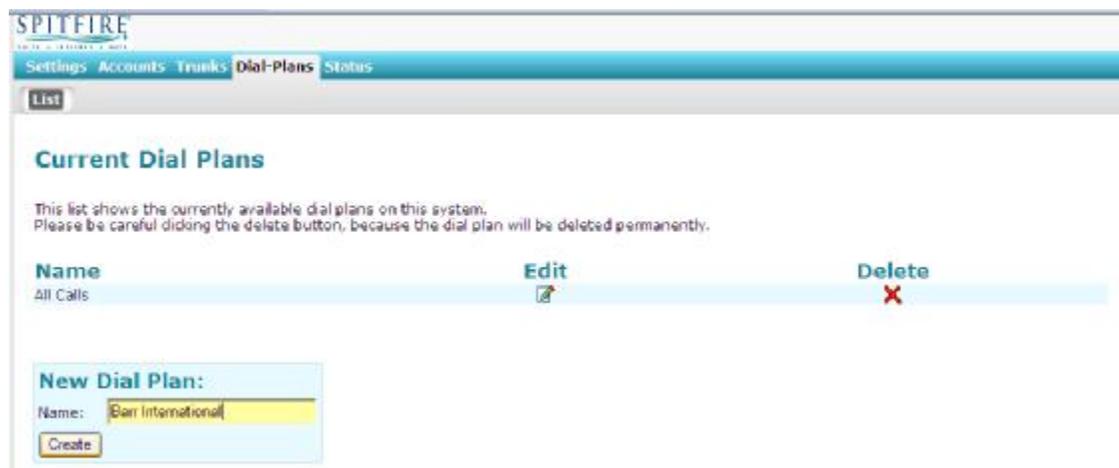
- Used to set default settings and timers for the domain.



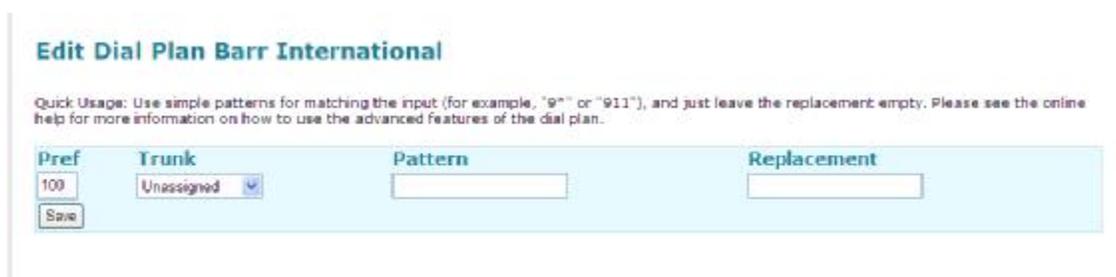
- Voicemail Timeout: Number of seconds that a call will ring at an extension before going to voicemail. *Users can over-ride this setting.*
- Voicemail Size: Number of messages that can be stored.
- Mailbox Escape Account: Where calls should be directed if caller dials "0" while listening to mailbox 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

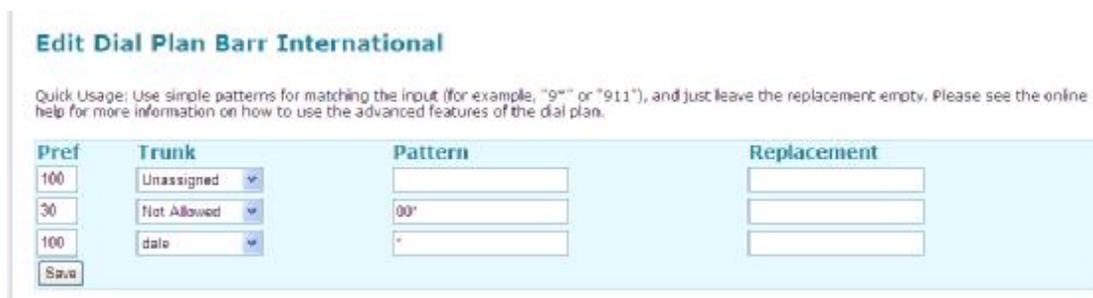
- Select Dial-Plans from the menu bar:
- Type in the name for the Dial Plan and click Create as shown below.



- 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.

Settings Accounts Trunks Dial-Plans Status Admin

List Create

Create New Accounts

Accounts are numbers that you can call on the PBX. Please select the account type that you would like to create.
Tip: You may enter more than one name. If you use a space between the names, the PBX will set up several accounts for you. If you use a slash between the names, you will set up one account with different alias names. For example, "123/theo 124/fred" will set up two accounts, the first with the names 123 and theo and the second with the names 124 and fred.

Account Type: Extension
 Dial Plan: Domain Default
 Plug and Play: Manual

| Number | First Name | Last Name | SIP Pass. | Web Pass. | PIN | Email |
|-----------------|------------|-----------|-----------|------------|------|-----------------|
| 1 2002/20759986 | Alison | Clark | Ytre78To | alison2002 | 8080 | alison.clark@te |

- 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.

Settings Accounts Trunks Dial-Plans Status

List Create

0000...Co5

| Account | Type (Name) | Status | Edit | Delete |
|-------------------|--------------------------------|--------|------|--------|
| 0000* | Extension (admin only) | | | |
| 1001 | Extension (main mailbox) | 2 Regs | | |
| 1010 | Paging (Page All) | | | |
| 2001 | Extension (Jack Frost) | 1 Regs | | |
| 2002 | Extension (Evie Powell) | 1 Regs | | |
| 2003 | Extension (Jeannette Anderson) | 1 Regs | | |
| 2004 | Extension (Virtual One) | | | |
| 2005 207141996 | Extension (Alison Clarke) | | | |
| 2070360015 <2005> | Auto Attendant (Main SA1) | | | |

Editing Extension 2005 207141996

| | |
|---|---|
| Account Number(s): | 2005 207141996 |
| Dial Plan: | Bar International |
| ANI: | |
| Send daily CDR (approx. 0.0): | |
| Show following ACC queues: | |
| First name (e.g. John): | Alison |
| Last name (e.g. Smith): | Clara |
| SIP Password: | ***** |
| SIP Password (repeat): | ***** |
| Web Password: | ***** |
| Web Password (repeat): | ***** |
| PIH (e.g. 1234): | ***** |
| PIH (repeat): | ***** |
| Timezone: | Default time zone |
| Default Language: | Default System VR Language |
| Web Language: | Default System Web Language |
| Upload picture: | <input type="text"/> <input type="button" value="Browse"/> |
| Wakeup time (e.g. 7:15): | |
| Watch the ends of the following extensions ("*" for all): | |
| Watch the presence of the following extensions ("*" for all): | |
| Watch following accounts on PAC: | |
| Explicitly specify park url: (url: ext:ext): | |
| Explicitly specify pickup preference: | |
| Permissions to monitor this account: | |
| Block outgoing caller-ID: | <input type="checkbox"/> No <input checked="" type="checkbox"/> 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 register and be operational.
- Click “System Information” and ensure that the account has registered.

SIP Identity
Status:
 Identity 2001@spitfire.spitfiretsp.net: Ok
 1 Status:
 Identity

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”

| Account | | Account 1 |
|------------------------------|---|-------------|
| Basic >> | | |
| Register Status | Registered | |
| Account Active | <input checked="" type="radio"/> On <input type="radio"/> Off | |
| Label | Ted | |
| Display Name | Ted | ? |
| Register Name | 2001 | ? |
| User Name | 2001 | ? |
| Password | | |
| SIP Server | spitfire.spitfiresp.net | Port 5060 ? |
| Enable Outbound Proxy Server | Enabled | ? |
| Outbound Proxy Server | pbx5.spitfiresp.net | Port 5060 |
| Transport | UDP | |
| Backup Outbound Proxy Server | | Port 5060 |
| NAT Traversal | Disabled | ? |
| STUN Server | 217.10.79.21 | Port 10000 |
| Voice Mail | 2001 | |
| Proxy Require | | ? |
| Anonymous Call | Off | |
| On Code | | |
| Off Code | | |
| Anonymous Call Rejection | Off | |
| On Code | | |
| Off Code | | |
| Missed call log | Enabled | |
| Auto Answer | Disabled | |
| Ring Type | common | |

- 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.

SPITFIRE
Settings Accounts Trunks Dial-Plans Status Admin

List Create **Edit**

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): 9030

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 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): 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.



The screenshot shows a web interface for configuring a mailbox. The title is "Redirection for Extension 1001". The interface includes a navigation bar with "Settings", "Accounts", "Trunks", "Dial-Plans", and "Status". Below the navigation bar, there are tabs for "List", "Create", "General", "Redirection", "Mailbox", "Email", "Registration", and "Permission". The "Redirection" tab is active. The configuration options are:

- Do not disturb: on off
- Agent Logged In: on off
- Incoming anonymous calls: No special treatment (dropdown menu)
- Hot Desking at: (text input)
- Call forward all calls to: (text input)
- Call forward calls when busy to: (text input)
- Call forward on no answer to: (text input)
- Call forward no answer timeout: 1 sec (dropdown menu)
- Call forward when not registered: (text input)

- 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.



The screenshot shows a web interface for creating new accounts. The title is "Create New Accounts". The interface includes a navigation bar with "Settings", "Accounts", "Trunks", "Dial-Plans", and "Status". Below the navigation bar, there are tabs for "List" and "Create". The "Create" tab is active. The form includes:

- Account Type: Auto Attendant (dropdown menu)
- Account Names (e.g. "123/f.feuerstein"): 2201 (text input)
- Create (button)

Below the form, there is a tip: "Tip: You may enter more than one name. If you use a space between the names, the PBX will set up several accounts for you. If you use a slash between the names, you will set up one account with different alias names. For example, '123/theo 124/fred' will set up two accounts, the first with the names 123 and theo and the second with the names 124 and fred."

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

PITFIRE
 Settings Accounts Trunks Dial-Plans Status
 List Create Edit IVR

Edit Attendant 2201 2075015678

Identity:
 Account Number(s): 2201 2075015678
 Name (e.g. Attendant 1): Sales

Behavior:
 Extension Input: When Extension Matches
 Say Name: No Yes
 Accounts that cannot be called:
 Accounts that may record a message: 2002 2005
 Dial Plan for outbound calls: All Calls
 ANI: 2075015678
 Send daily CDR report to:
 Set Language: Keep selected language
 Second Language: Only one language
 Permissions to monitor this account:

Timeout Handling:
 Redirect Number: 2001
 Timeout (s): 6
 Hangup Timeout:

Night Service:
 Service Flag Account: 9000
 Night Service Number: 1001

Dial By Name:
 Input that triggers name search:
 Start Search: After 1 Digit Input

Direct Destinations:

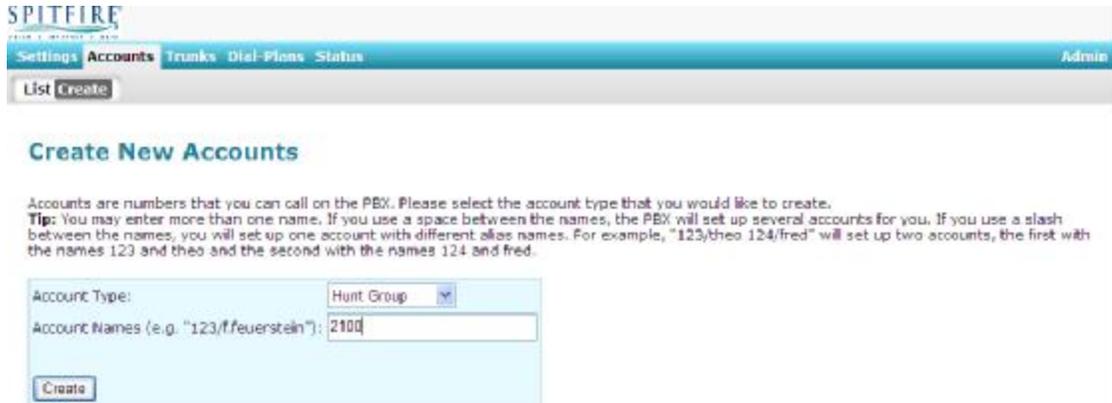
| User Input: | | Destination: |
|-------------|-------------|--------------|
| 1 | No playback | 2001 |
| 3 | No playback | 2001 |
| | No playback | |

Save

- 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.

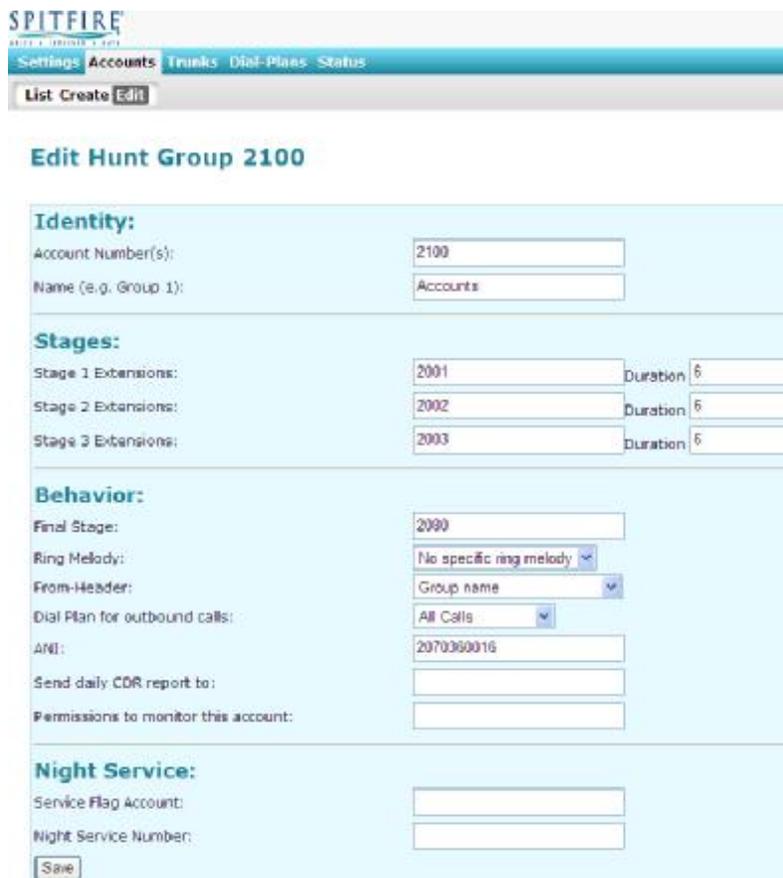
10. Configuring Hunt Groups

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



The screenshot shows the SPITFIRE web interface. The top navigation bar includes 'Settings', 'Accounts', 'Trunks', 'Dial-Plans', and 'Status'. The 'Accounts' tab is active, and the 'Create' button is highlighted. Below the navigation bar, the page title is 'Create New Accounts'. A tip explains that accounts are numbers on the PBX and that multiple names can be entered for a single account using slashes. The form contains an 'Account Type' dropdown menu set to 'Hunt Group' and an 'Account Names' text input field containing '2100'. A 'Create' button is located at the bottom left of the form.

- Select List under Accounts.
- Select Edit next to the Hunt Group.



The screenshot shows the SPITFIRE web interface with the 'Accounts' tab active and the 'Edit' button highlighted. The page title is 'Edit Hunt Group 2100'. The form is divided into several sections: 'Identity' with fields for 'Account Number(s)' (2100) and 'Name (e.g. Group 1)' (Accounts); 'Stages' with three rows for 'Stage 1 Extensions', 'Stage 2 Extensions', and 'Stage 3 Extensions', each with a 'Duration' of 6; 'Behavior' with fields for 'Final Stage' (2090), 'Ring Melody' (No specific ring melody), 'From-Header' (Group name), 'Dial Plan for outbound calls' (All Calls), 'ANI' (2070360016), 'Send daily CDR report to:', and 'Permissions to monitor this account:'. The 'Night Service' section includes 'Service Flag Account:' and 'Night Service Number:' fields. A 'Save' button is located at the bottom left of the form.

- 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.

The screenshot shows the SPTIFIRE web interface. At the top, there are navigation tabs: Settings, Accounts, Trunks, Dial-Plans, and Status. Below the navigation, there are buttons for 'List' and 'Create'. The main heading is 'Create New Accounts'. Below this, there is a tip: 'Accounts are numbers that you can call on the PBX. Please select the account type that you would like to create. Tip: You may enter more than one name. If you use a space between the names, the PBX will set up several accounts for you. If you use a comma between the names, you will set up one account with different alias names. For example, "123/theo 124/fred" will set up two accounts, the names 123 and theo and the second with the names 124 and fred.' Below the tip, there is a form with two fields: 'Account Type:' with a dropdown menu set to 'Conference', and 'Account Names (e.g. "123/f.feuerstein"):' with a text input field containing '1020'. There is a 'Create' button at the bottom of the form.

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

The screenshot shows the SPTIFIRE web interface. At the top, there are navigation tabs: Settings, Accounts, Trunks, Dial-Plans, and Status. Below the navigation, there are buttons for 'List', 'Create', and 'Edit'. The main heading is 'Edit Conference 1020'. Below this, there is a form with several sections: 'Identity:' with 'Account Number(s):' set to '1020 2070360019' and 'ANI:' empty; 'Authentication:' with 'Conference Name:' set to 'Weekly Sales', 'Mode:' set to 'Ad-hoc conference', 'IVR Language:' set to 'Default System IVR Language', 'Before entering the conference:' set to 'Just play a short tone', 'Moderator Access Code:' empty, 'Participant Access Code:' set to '1234', and 'Permissions to monitor this account:' empty. There is a 'Save' button at the bottom of the form.

- 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”.

SPITFIRE
VOICE & INTERNET DATA

Settings Accounts Trunks Dial-Plans Status Admin

List Create

Create New Accounts

Accounts are numbers that you can call on the PBX. Please select the account type that you would like to create.
Tip: You may enter more than one name. If you use a space between the names, the PBX will set up several accounts for you. If you use a slash between the names, you will set up one account with different alias names. For example, "123/theo 124/fred" will set up two accounts, the first with the names 123 and theo and the second with the names 124 and fred.

Account Type: Calling Card

Account Names (e.g. "123/f.euerstein"): 100

Create

- 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”

SPITFIRE
VOICE & INTERNET DATA

Settings Accounts Trunks Dial-Plans Status Admin

List Create Edit

Edit Calling Card 100

Identity:

Account Number(s): 100

Behavior:

Dial Plan: all

ANI: 20760812345

Send daily CDR report to: joe.bolggs@company.co.uk

Callback: on off

Caller-ID: Show Block

Allow calling accounts on the PBX: Allow Don't Allow

Entering telephone numbers: User must press pound

Read out balance when lower than (e.g. "1.00"):

Permissions to monitor this account:

- 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.

The screenshot shows the SPITFIRE web interface. At the top, there's a navigation bar with 'Settings', 'Accounts', 'Trunks', 'Dial-Plans', 'Status', and 'Admin'. Below that, a sub-menu includes 'List', 'Create', 'General', 'Redirection', 'Mailbox', 'Email', 'Registration', and 'Permission'. The main heading is 'Edit Extension 3016 2070360016'. A paragraph explains that permissions should be given only to persons who can change passwords, create/delete accounts, change dial plans, and so forth. Below this, there are two sections: 'Administration Permissions' and 'Call Permissions'. In 'Administration Permissions', 'Domain Administrator' is set to 'off'. In 'Call Permissions', there are five input fields, each containing '3017', corresponding to: 'Barge into calls of the following extensions:', 'Teaching (whisper mode) for the following extensions:', 'Listen in to calls of the following extensions:', 'Call the following extensions even if DND is set:', and 'Intercom to the following extensions:'. A 'Save' button is located at the bottom left of the form area.

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.

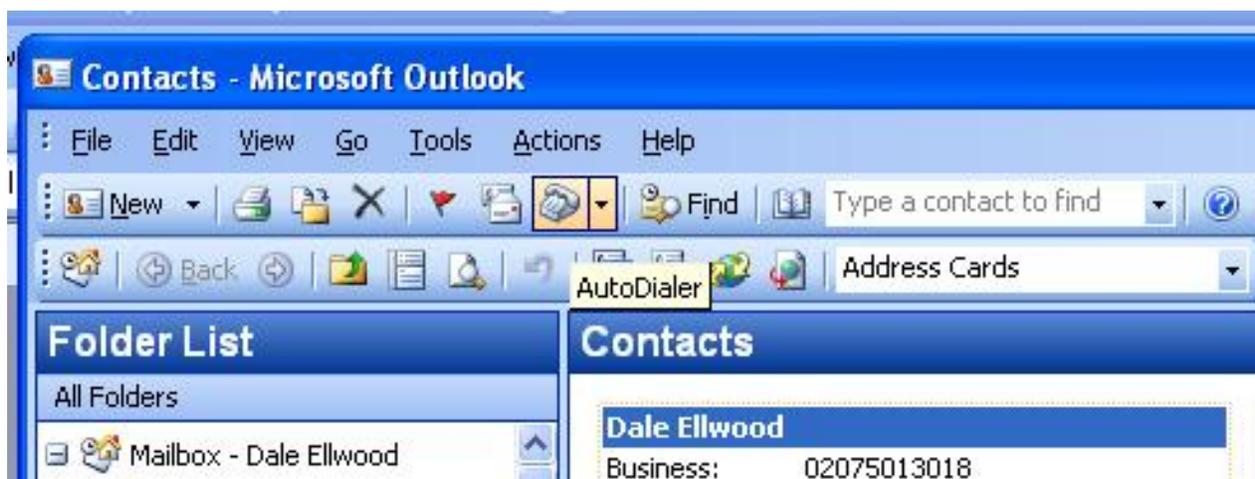


Figure 1

- On the “New Call” screen, click “Dialing Options”

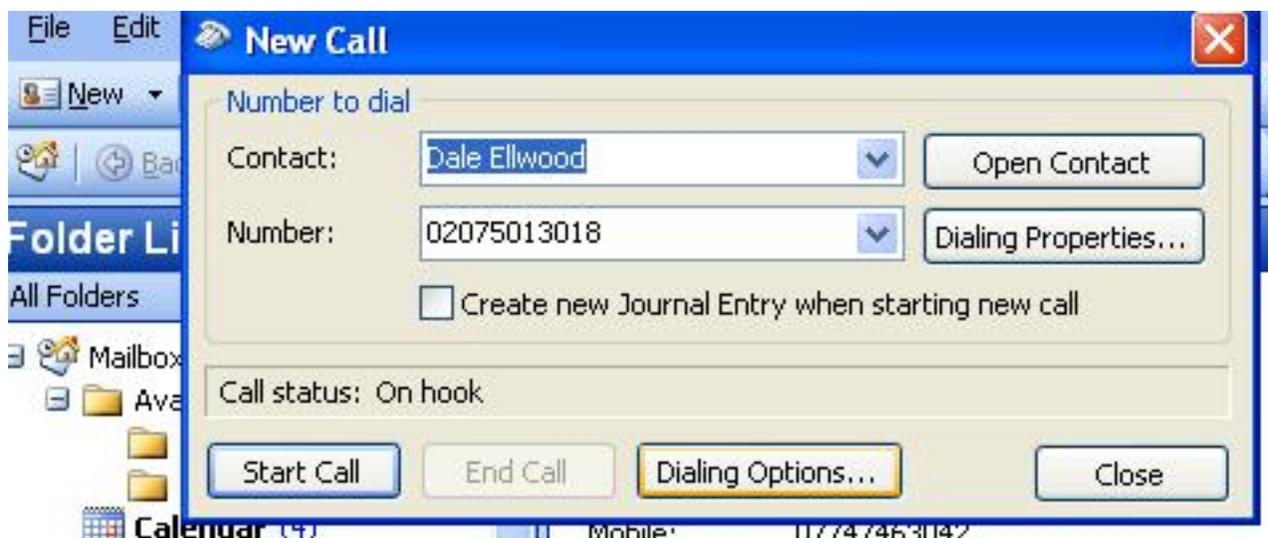


Figure 2

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



Figure 3

- Click on “Line Properties”

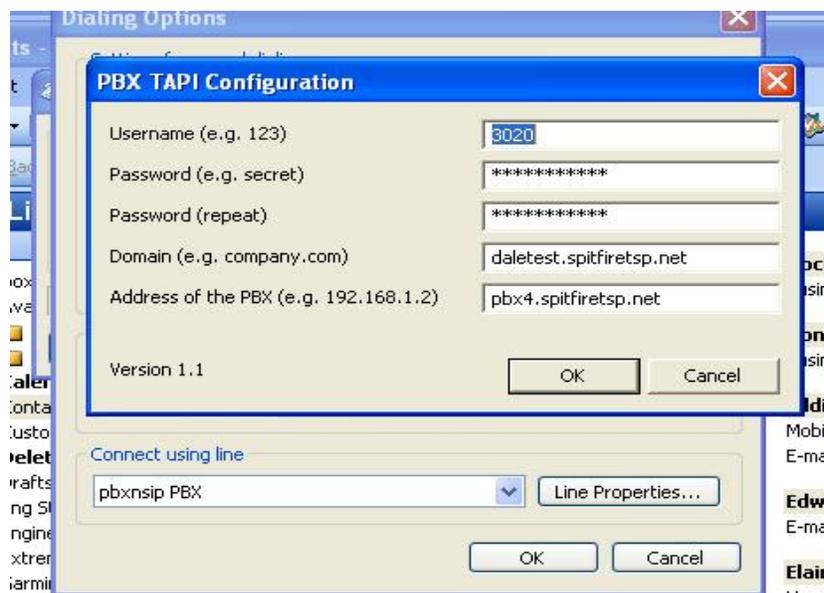


Figure 4

- 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.

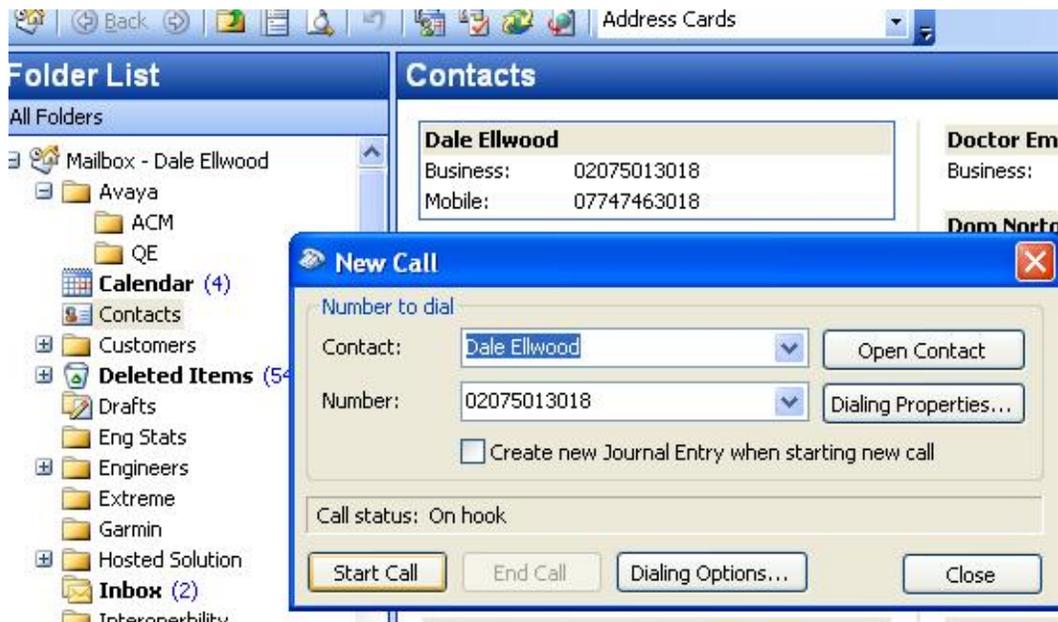


Figure 5

- 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. |

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| 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. |

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| 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. |
| Twining | 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. |