# TELSTRA BUSINESS SIP® PORTAL ADMINISTRATOR GUIDE

# Contents

# 1. Introduction

Welcome to the Telstra Business SIP® Customer Management Portal.

Using a web browser, from a central location administrators can monitor and manage other administrators, site settings and end users.

The Portal is used for the following actions:

- Retrieving device credentials for OneAccess devices (SIP NTU, One100 IAD) needed for initial installation.
- Activating encryption over the segment: Telstra devices -> Telstra core, where the broadband access product traverses untrusted networks, e.g. internet.
- Performing number migration from the old ISDN/PSTN service to the new Business SIP service.
- Assigning purchased feature packs to individual numbers
- Performing day-to-day management of features described in this document.

This guide also details service migration, referred to in the Customer Management Portal as 'Migrations' – the feature that enables self-migration of existing services that have been configured through a sales process. More about this in Section 3.15 Migrations.

## 1.1. Browser compatibility

The Customer Management Portal is designed to be compatible with the following browsers:

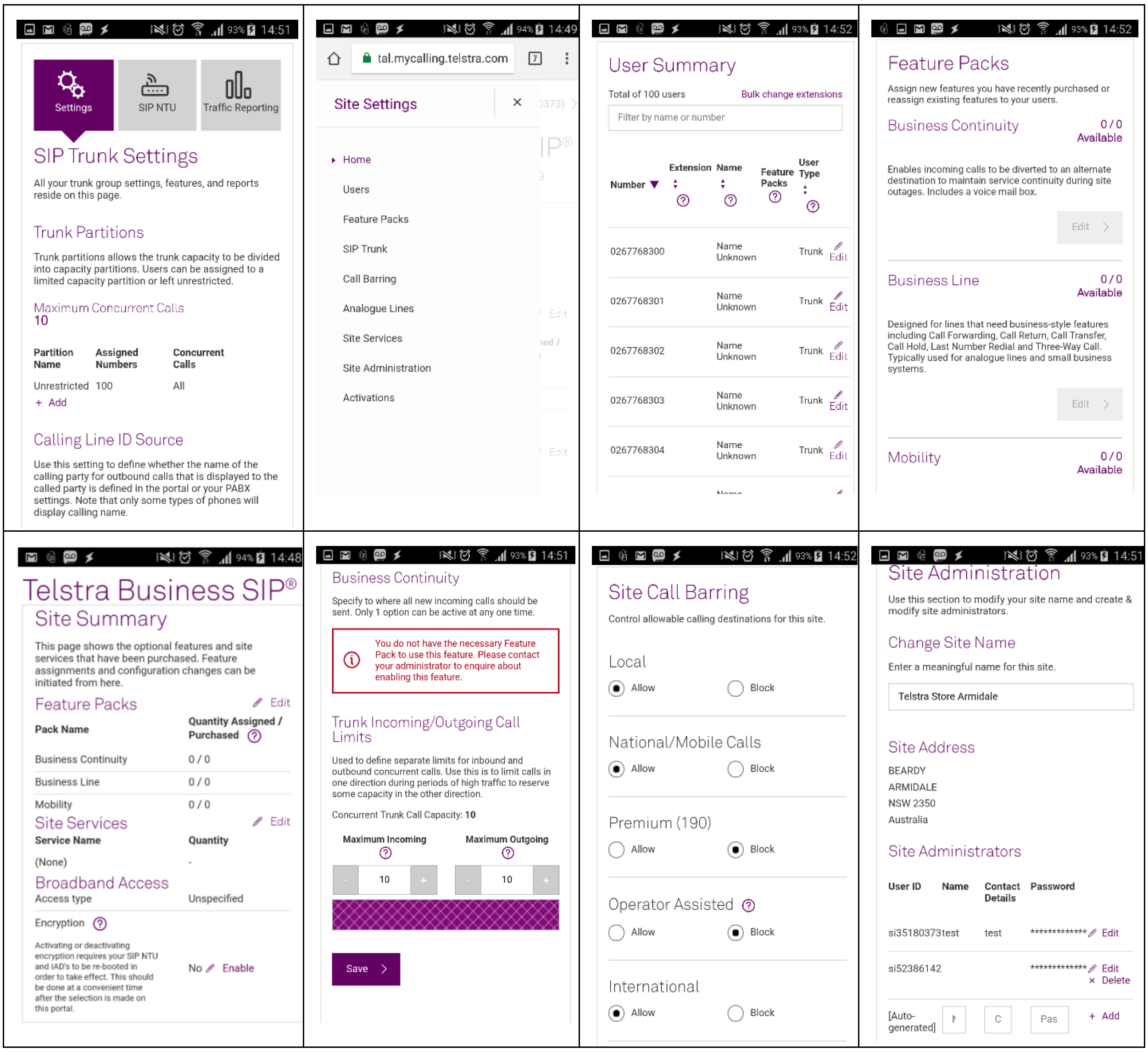
- Firefox
- Google Chrome for desktop.

## 1.2. Mobile device compatibility

The Customer Management Portal is optimised for mobile devices and is a convenient way of managing your Business SIP product from almost anywhere.

Open a supported browser on your mobile device, enter the url and log in using your Business SIP Portal Username and Password.

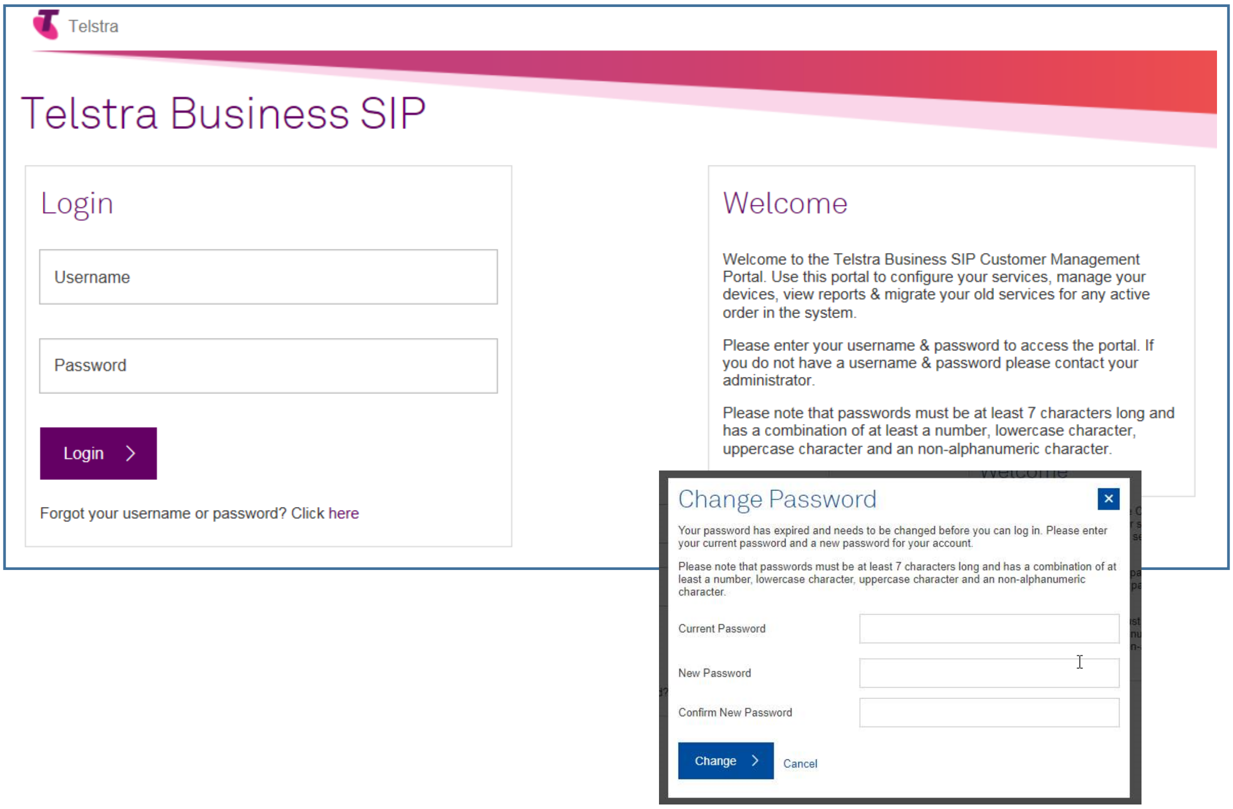
Below are some sample images:



## 1.3. Log in

To access the Customer Management Portal, https://portal.mycalling.telstra.com, log in with your username (starting with ‘cu’ or ‘si’) and administrator password. Username/passwords are supplied by email to the customer administrator at the time of service establishment.

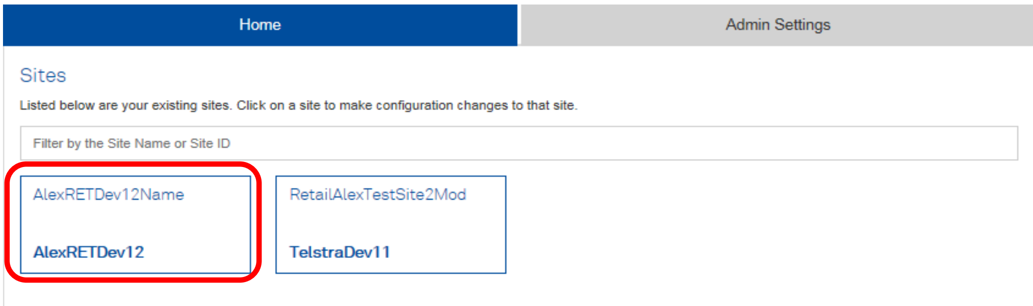
On the very first log in by the Administrator, the password reset pop-up will appear as shown below. Enter the current password then the new password, confirm the new password and click Save. (Your password must be at least seven characters long and contain at least one uppercase letter, number and special character).



# 2. Customer level administration

The Customer Administration login provides management of Administrator Users across multiple Sites for a given Customer/Business; however all Business SIP feature configurations can be performed equally through Customer or Site Administration access. The main difference for Site Administrators, is that they will only ever see their local Site, which is controlled by their login Username and Password.

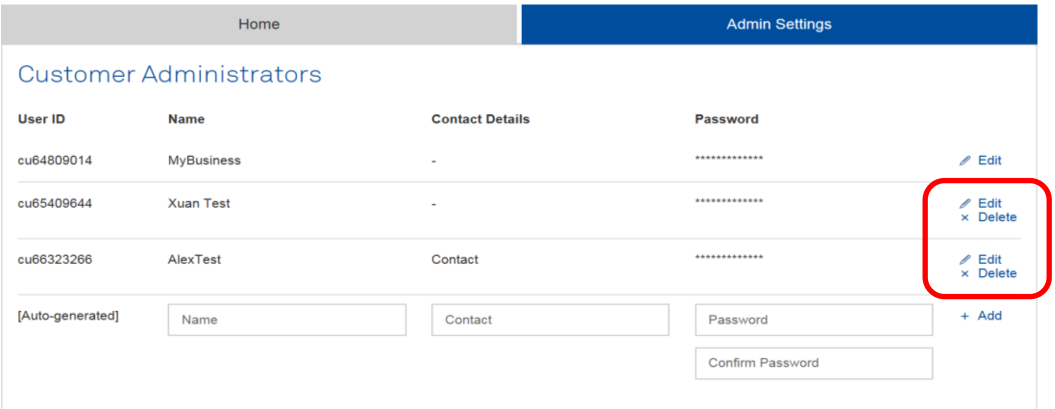
Once logged in, the Customer Administrator will see all their Sites on the 'Home' tab (shown) and can access the Administrator accounts via the 'Admin' tab, from where Customer Administrator passwords can be managed.



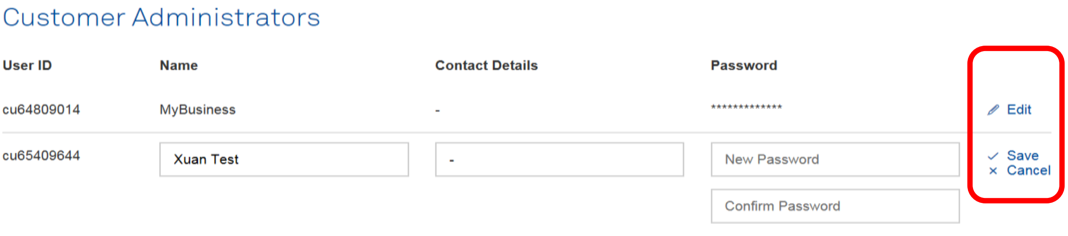

## 2.1. Create, edit and remove customer administrators

Select the **Admin Settings** tab to open the page that allows you to create, edit and remove customer administrators and reset/change passwords.

Note: The first customer administrator can be edited but not deleted.




For subsequent administrators, click × Delete to remove an existing administrator or the Edit icon, which will present the edit fields as shown below:




Once you've finished making changes, click the ✓ Save icon or × Cancel.
To create a new administrator click + Add.

# 3. Site level Administration

All the Telstra Business SIP services can be configured at the site level, including **feature packs** and **site services** (additional to the basic services and purchased separately).

**Feature packs** provide users with specific functionality as they answer or make calls.

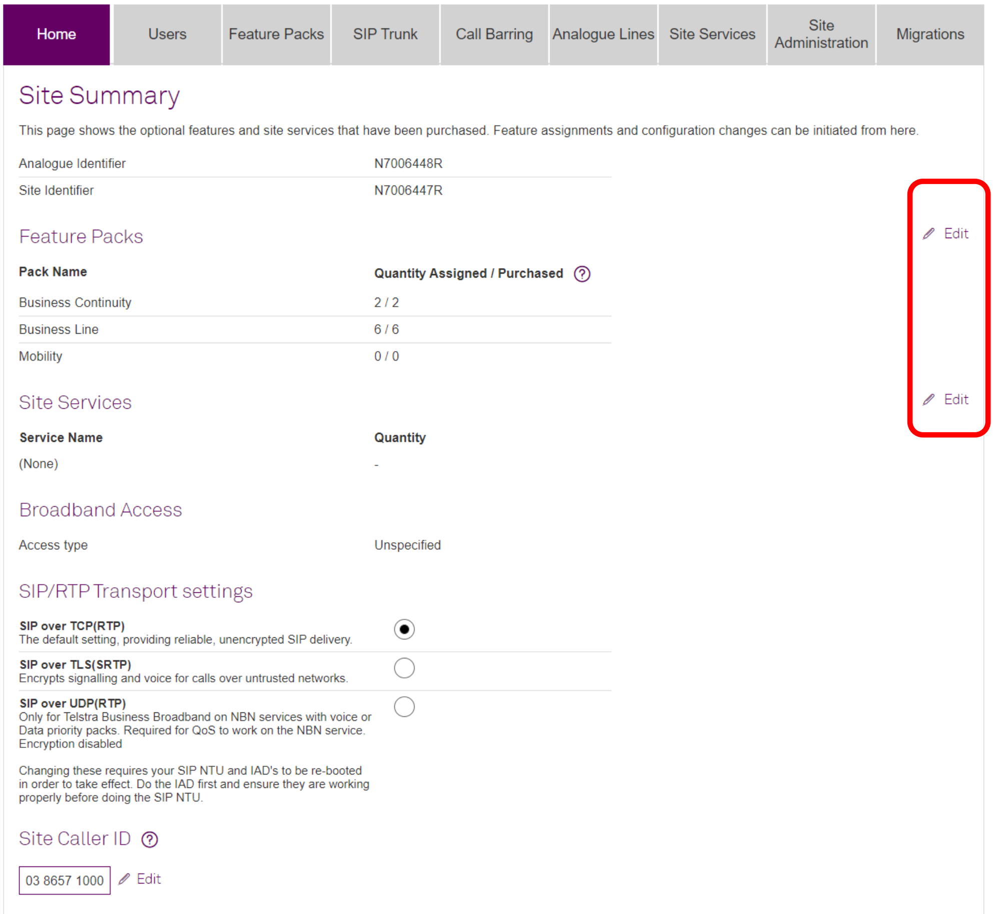
**Site services** are configured to manage how incoming calls are treated. Site services provide control and efficiency by applying call-handling rules and automation through a self-service capability extended to callers in the form of an IVR (interactive voice response). See virtual receptionist and hunt groups for details.

**Broadband Access** presents the access type and information about the service.

## 3.1. Site Summary

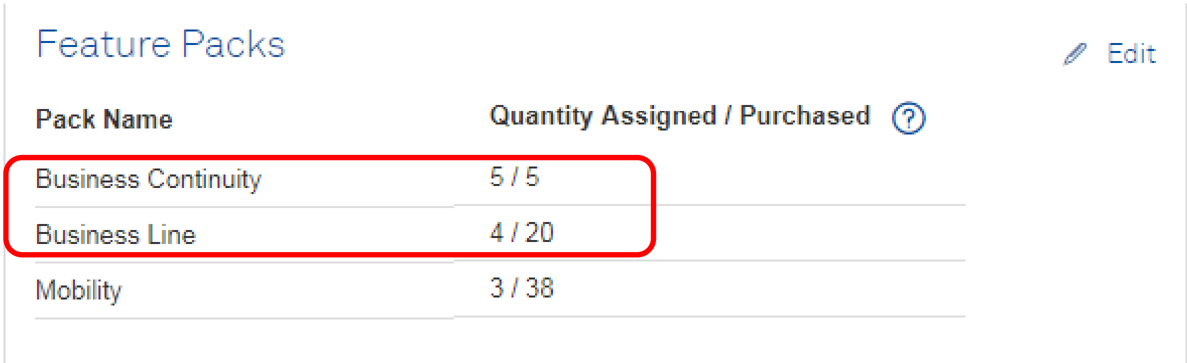
At the Site Summary level you can see at a glance.

The Feature Packs and Site Services present their quantities and current allocations and the Edit icon alongside **Feature Packs** and **Site Services** takes you directly to the respective tab where you can make changes. These tabs and others are covered in detail in the sections below.



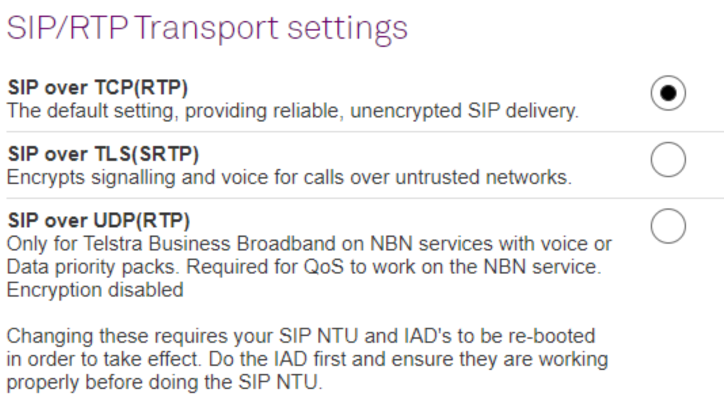
Against each feature pack the numbers represent the available instances not yet assigned (on the left) and the total quantity available (on the right).

The example below shows that out of five available '**Business Continuity'** feature packs there are none unassigned, whereas the '**Business Line'** shows only 4 assigned out of 20.





Hunt Group and Virtual Receptionist Site Services can also be ordered and are detailed in sections 3.12 & 3.13.




SIP/RTP Transport Settings provides 3 options depending on site requirements.

SIP over TCP (RTP) is the default option and, in most cases, can be left as-is.

SIP over TLS (SRTP) can be employed where an untrusted broadband provider is in place, where there may be risk of unauthorised personnel accessing calls. It can also help to traverse routers/firewalls not optimised for SIP, e.g., where SIP ALG would otherwise block the signalling messages.

SIP over UDP (RTP) is used where Telstra Business Broadband Voice or Data Priority Packs have been purchased and a Telstra Gateway Pro (V7610) is in place (see picture below).

This device will not take advantage of the priority packs unless SIP over UDP has been activated.

The priority packs activate Traffic Class 2 (TC2) bandwidth on NBN and, if a BYO router is in place, it will need it's settings adjusted to take advantage of the TC2 bandwidth – this may or may not also require SIP over UDP. Please enquire with your router maintainer.

Changing the SIP/RTP settings makes changes to the SIP NTU and/or IAD settings. For these settings to activate, the devices will need to be power-cycled, as per the note on the portal.


Site Caller ID allows one of the assigned phone numbers to be allocated as the "Site Caller ID". This is particularly useful for analogue lines which are tied to a specific phone number, where an advertised number would be preferable as the caller ID displayed to the called person. Using this feature allows analogue lines to display the Site Caller ID when outbound calls are made. This also works for numbers assigned to the SIP trunk.

Modify the "Site Caller ID" by clicking on "Edit" and choosing the desired number. Then navigate to the Users/Outgoing Calls tab (see section 4.2) and enable the "Use Site Caller ID" feature for each user that needs the change made. The "Outbound Caller ID" field will then display the number that will be used as CLI for outbound calls.

Note that only one number at a time can be assigned as the Site Caller ID.

## 3.2. Customer Management Portal feature quick links

This list maps Customer Management Portal features to the sections in this guide and includes hyperlinks for ease of navigation. The right column presents the links as they appear in the Customer Management Portal tab labels to help with familiarisation.

| **Feature/action** | **Navigation to Customer Management Portal section** |
|---|---|
| Change **Customer** Administrator password - create/delete/edit | Admin tab |
| Change **Site** Administrator password create/delete/edit | Site Administration tab |
| Change **User** password/edit user | Users tab > User summary |
| Create extension shortcuts | Users tab> Bulk extension change |
| Make bulk extension change, based on phone number | Users tab> Bulk extension change |
| Make bulk extension change, based on sequence | Users tab> Bulk extension change |
| Enable feature packs | Feature packs tab |
| Assign features to users | Feature packs tab |
| Set up trunk partition | SIP trunk tab > Settings > Trunk partitions |
| Edit trunk partition – name, max calls, phone numbers | SIP trunk tab > Settings > Trunk partitions |
| Set up/configure calling line ID source | SIP trunk tab > Settings > Trunk partitions |
| Set up/configure business continuity | SIP trunk tab > Settings > Trunk partitions |
| Set up/configure trunk incoming/outgoing call limits | SIP trunk tab > Settings > Trunk partitions |
| Change customer premises equipment (CPE) device/get network termination unit (NTU) device credentials | SIP Trunk tab > SIP NTU |
| Traffic reporting | SIP trunk tab > Traffic reporting |
| Set up site call barring | Call barring tab > |
| Set up analogue lines<br>Change analogue equipment (CPE) device(s) /get credentials. | Analogue lines tab > |
| Set up hunt groups | Site services tab > |
| Set up auto-attendant | Site services tab > |
| Migration | Migrations tab > |

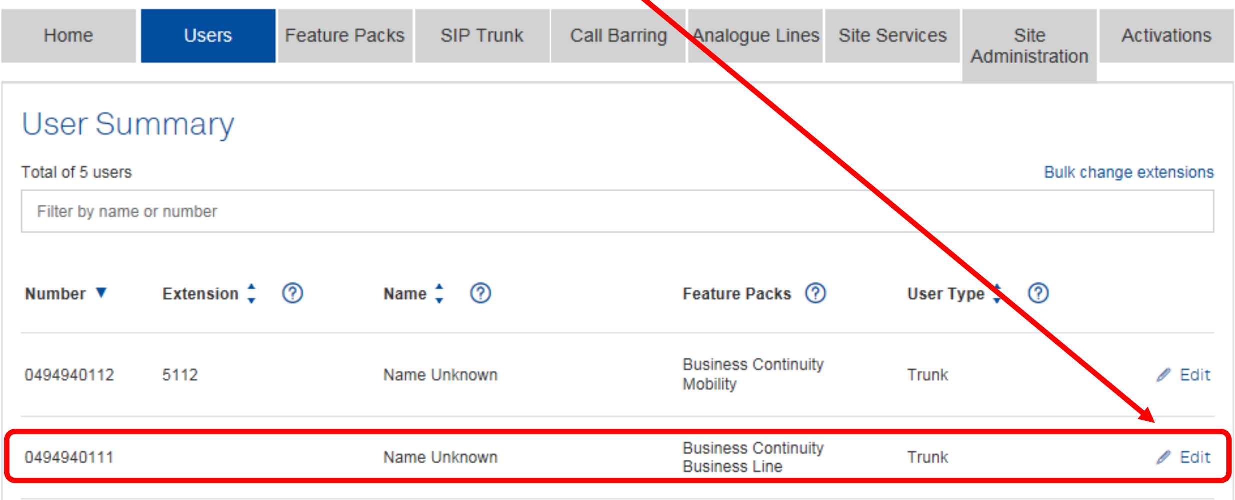
**Note:** Portal Help information is available in section 5 on page 68.

## 3.3. How to change a user’s name/password

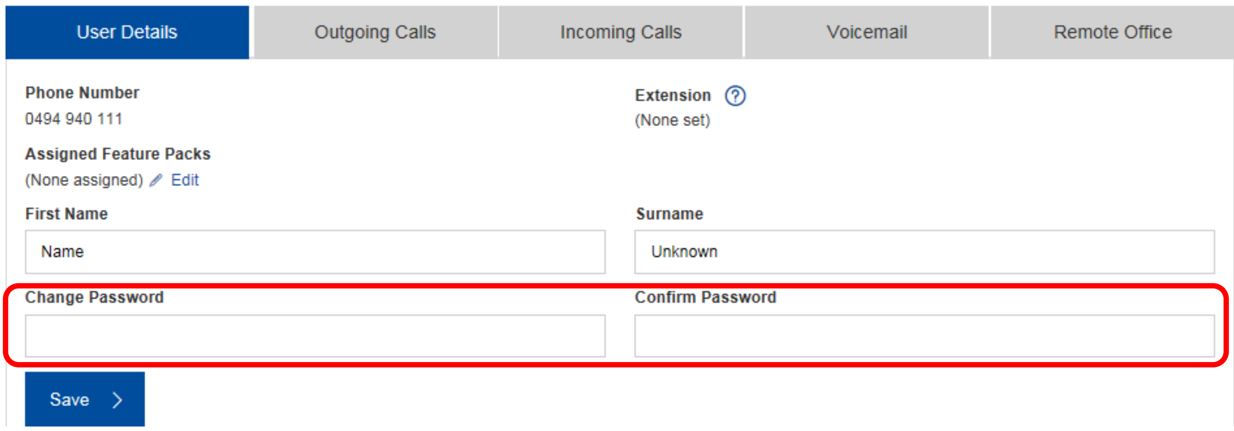
The users tab presents all the Full National Numbers (FNNs) associated with your business. This is where you can access the details for trunk and analogue users.

To change the name and passwords for users:

1. Log in to the Customer Management Portal using your administrator username and password.
2. Select the users tab to view the user summary.
3. From the list of users, click Edit to the right of that user. This will open the user details page.




4. In the name and password fields, enter the details as required. Remember, your password is at least seven characters long and contains at least one uppercase letter, number and special character.



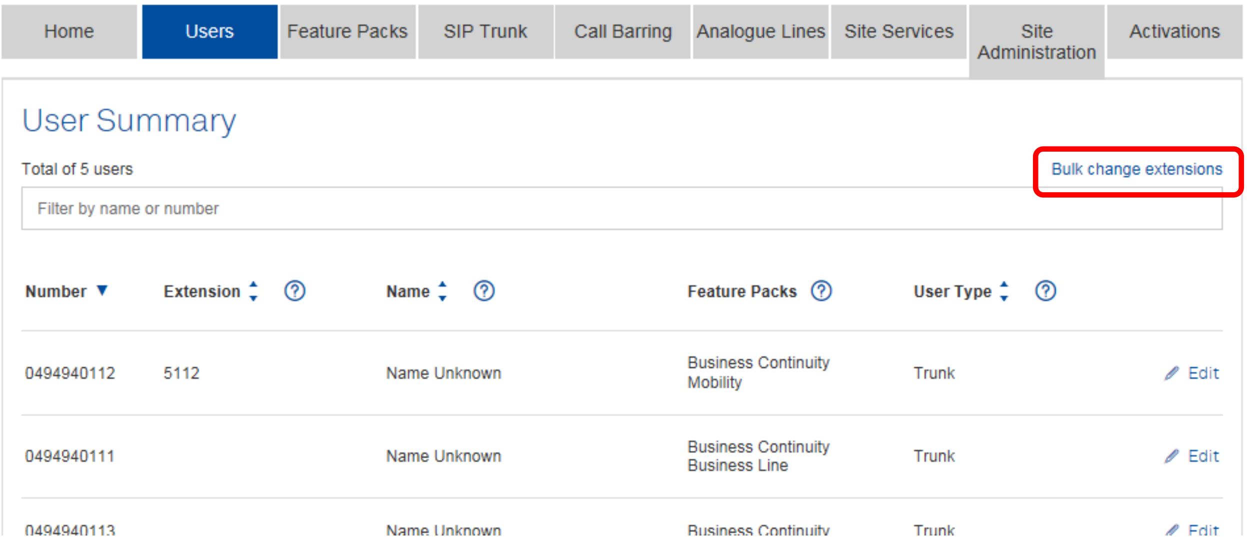

5. Click Save.

**Note:** From this view, you can also make changes to that user’s outgoing calls; incoming calls; voicemail; and remote office. Refer to the User features (section 4) for details.

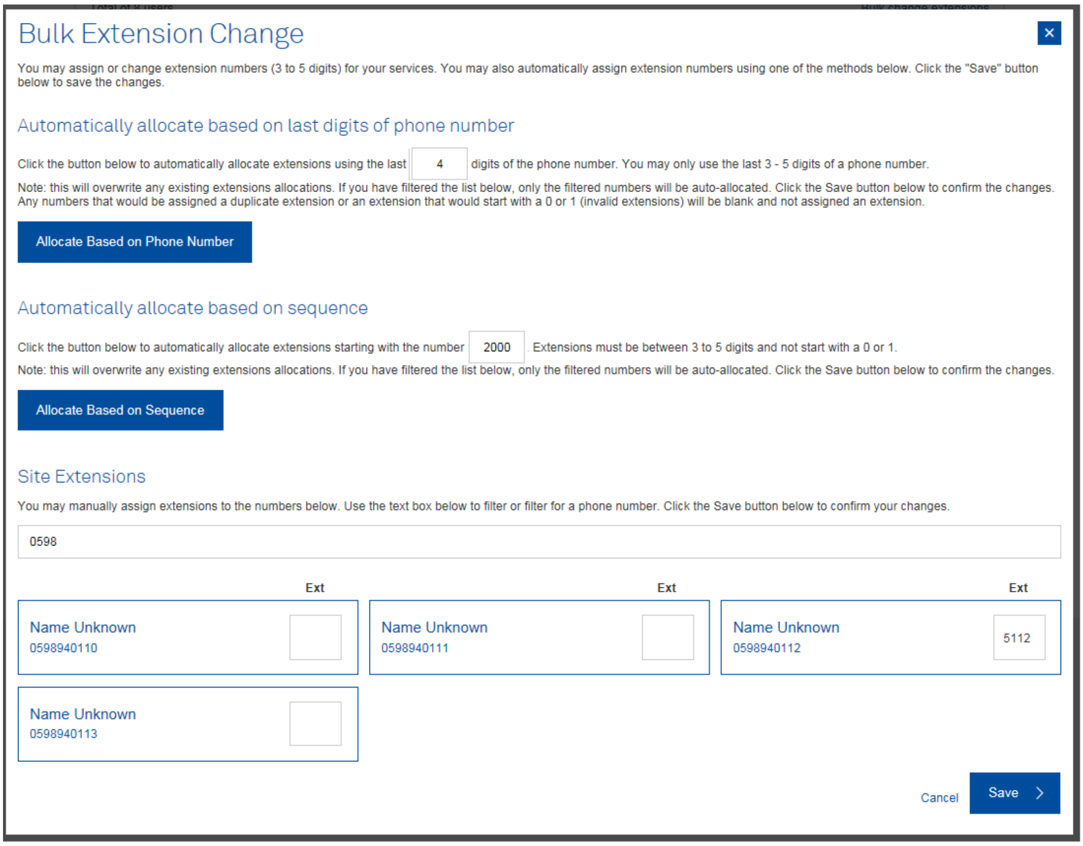
## 3.4. Create extension shortcuts

You can create shortcut numbers for users to call each other using only 3, 4 or 5 digits. There are some rules around what numbers can and can't be set as shortcuts, such as shortcuts cannot begin with either '0' or '1'. A quick way to assign numbers to all users is to use bulk change extensions.

To do this select the **Users** tab (**User Summary**) and click on **Bulk change extensions** as shown below.




Clicking **Bulk change extensions** opens the following pop-up window:



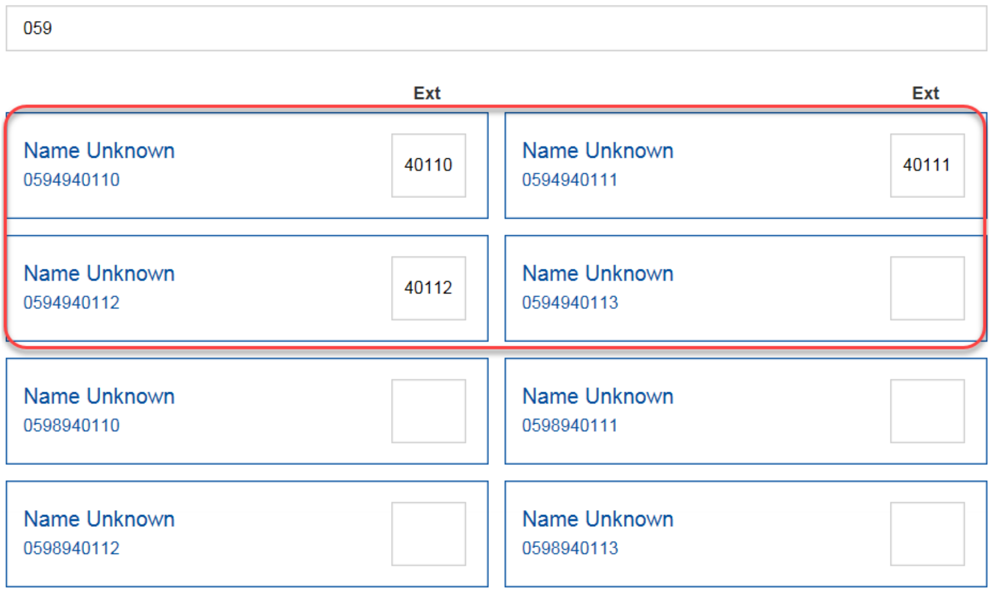
Extension shortcuts can be created in three different ways:

1. Manually. This can take considerable time depending on the number of extensions.
2. Automatically based on phone number. This is very quick as the shortcuts are based on the existing number. Note: number ranges that include 1s or 0s in the third, fourth of fifth character positions may fail as extensions cannot start with 0 or 1.
3. Automatically based on sequence. This is also very quick as the shortcuts are based on the starting number you choose, remembering that extension numbers cannot start with 0 or 1.

### 3.4.1. Manual allocation

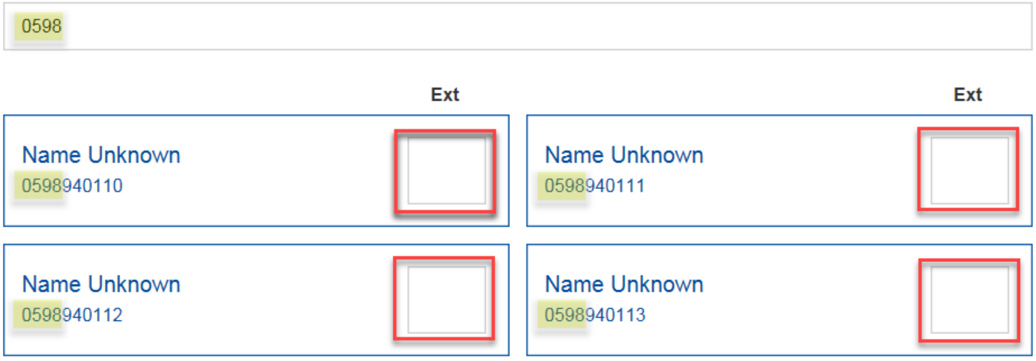
You can manually assign extensions to numbers. Click save to confirm your changes.

**Note:** The filter is dynamic and will immediately start to filter as you enter the number as shown in the sample below.




The list above shows the full list of numbers containing **059** (including **0594n** and **0598n** numbers) and the list below shows the filtered group of numbers containing **059<u>8</u>** only, i.e. the group circled in red above has been eliminated.

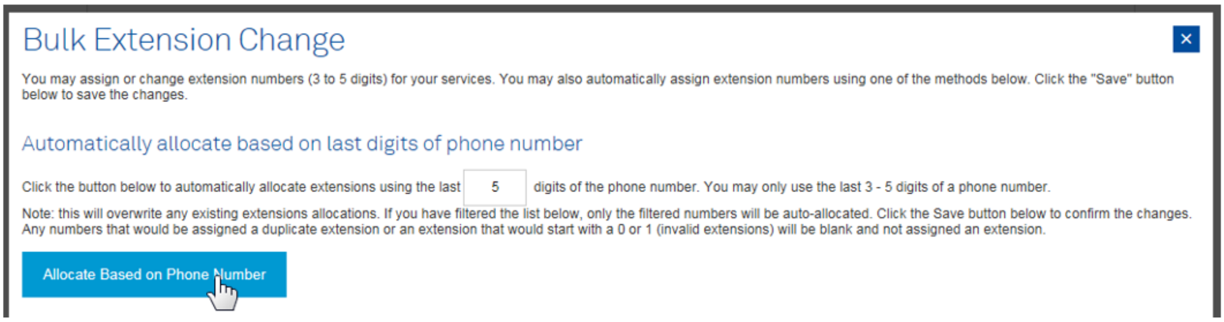
The extension shortcuts can now be entered in the empty boxes. In this example both groups are clearly visible/accessible in the same view; where this is not the case a scroll bar appear at right of the window.



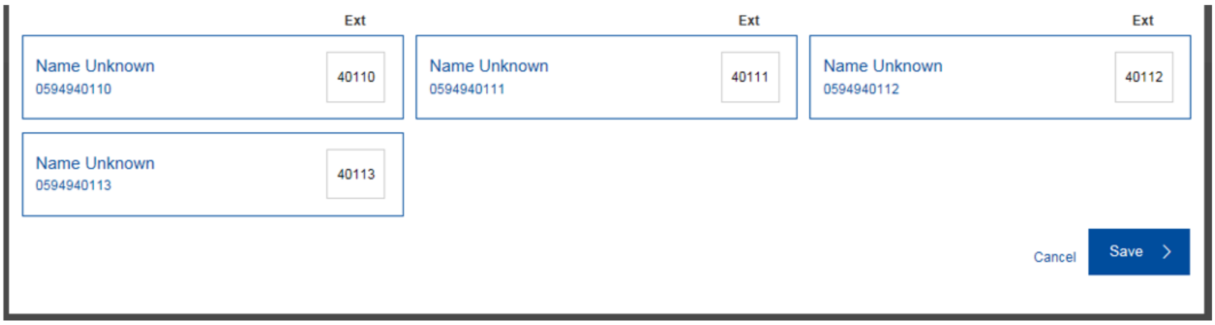

### 3.4.2. Allocation based on phone number

The examples below show the difference between the phone number and sequence options.

Given the rule that states no extension can start with a 0 or 1, it may mean that some extensions require manual allocation. This all depends on your particular number configuration.



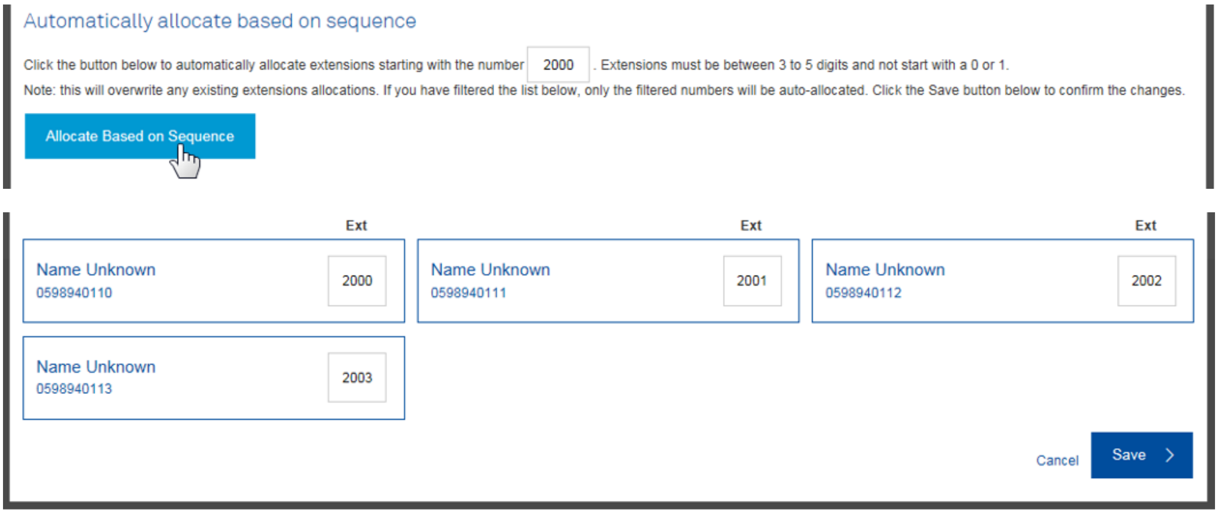

In this example, with the numbers shown below the extensions starting with 4 are valid in the 5-digit scenario, but choosing a 4-digit extension means that the extensions all start with ‘0’, which is invalid. The result is that all the ‘Ext’ boxes remain blank.




Click Save.

### 3.4.3. Allocation based on sequence

Entering a starting number in the field shown in the screen capture below sequences the numbers for the users selected. Using this method means prevents allocation issues.




Click Save.

## 3.5. Feature packs

Feature packs help users manage their calls and work effectively and efficiently.

There are three types of feature packs.

1. Business continuity
2. Business line
3. Mobility

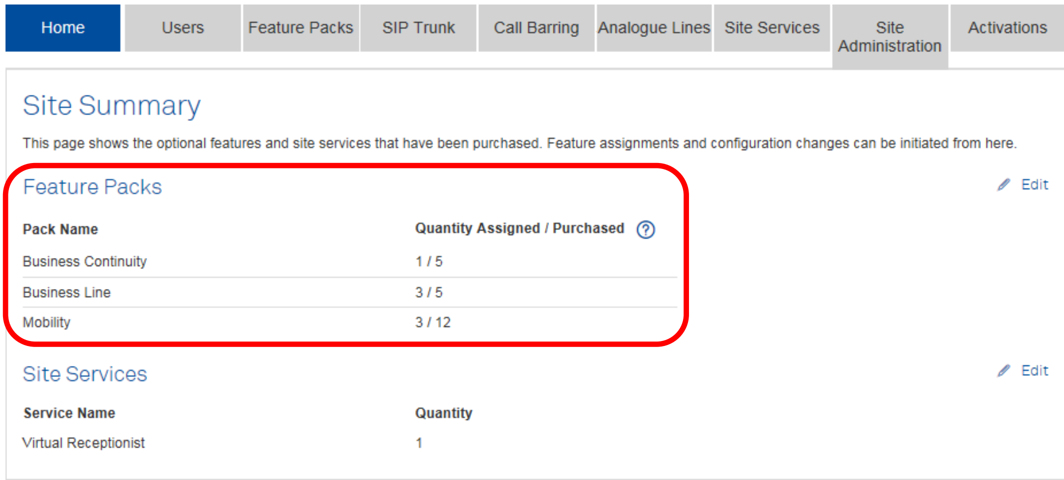
**Business continuity** – enables incoming calls to be diverted to an alternative destination during site outages (when calls cannot be received at the primary business location due to fire, flood or other such emergency that means that no calls can be received or answered).

**Business line** –designed for lines that need business-style features including call forwarding, call return[#], call transfer[*], call hold[*], last number redial[*] and three-way call[*].
Typically used for analogue lines and small business systems.

**Mobility –** enables mobile users to continue to receive calls when away from the office. Includes SIM ring, remote office and call forwarding busy/no answer/not reachable.

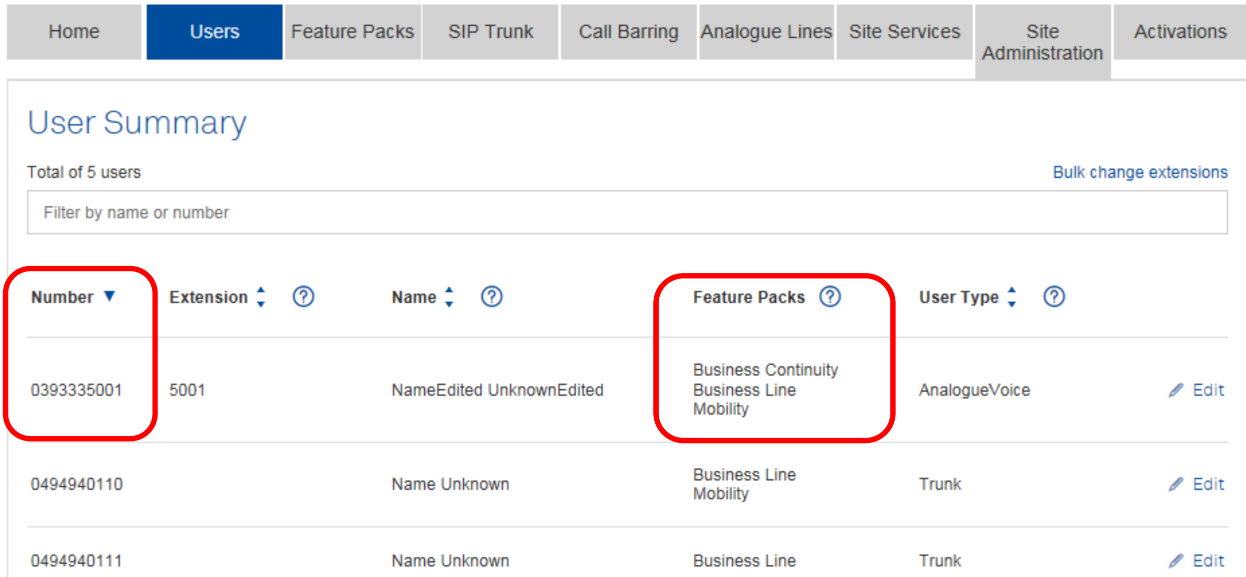
Feature packs are available to users who are assigned to them, as opposed to the other way around. This is because the feature packs are purchased against the site in specific quantities.

View how many feature packs are available to the site from the **Home** tab on the **Site Summary** page.

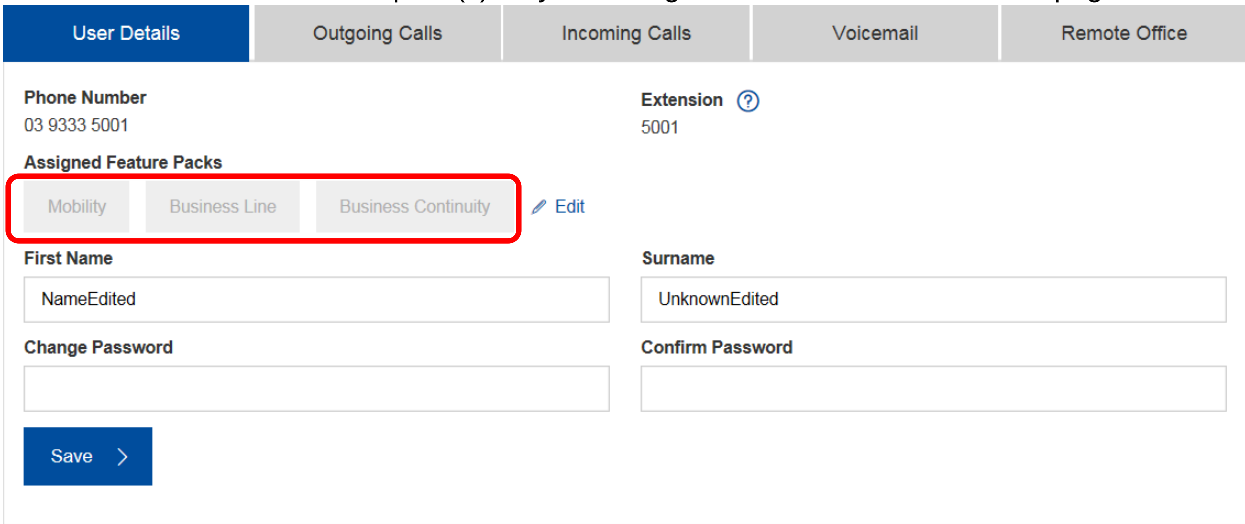



*Features provided in feature packs but which have no configuration in the Customer Management Portal.

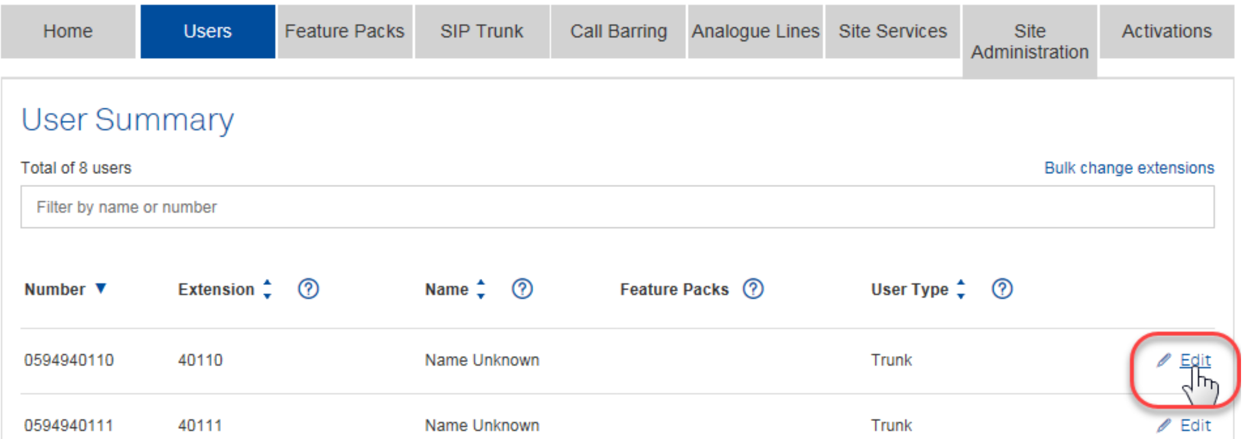
You can also see how the feature packs are assigned at the user level from the **Users** tab.



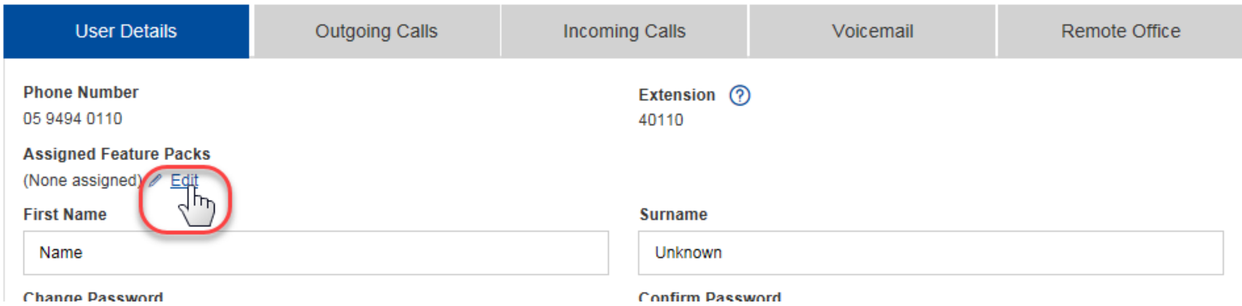
Users can view which feature pack(s) they are assigned to on their **User Details** page.




Assign numbers/users to feature packs via the **Feature Packs** tab or via the **Users** tab using the Edit button as shown below.




Look for **Assigned Feature Packs** and select Edit.




This opens the page that allows you to see the numbers/users who have the feature pack assigned and how many are available.

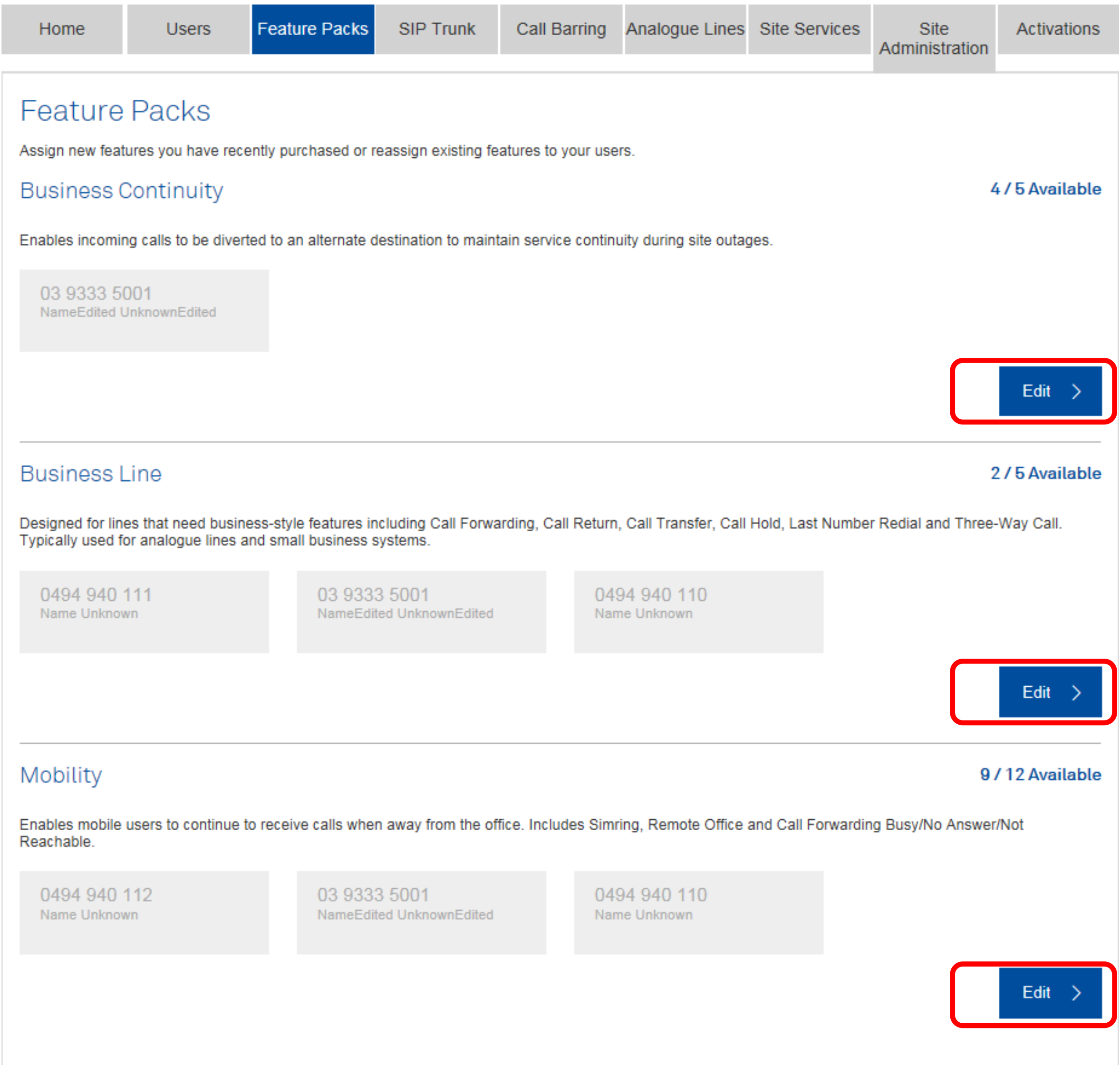
### 3.5.1. View feature packs assignments

You can view the current Feature Pack assignments two ways:

- By feature pack
- By user/number

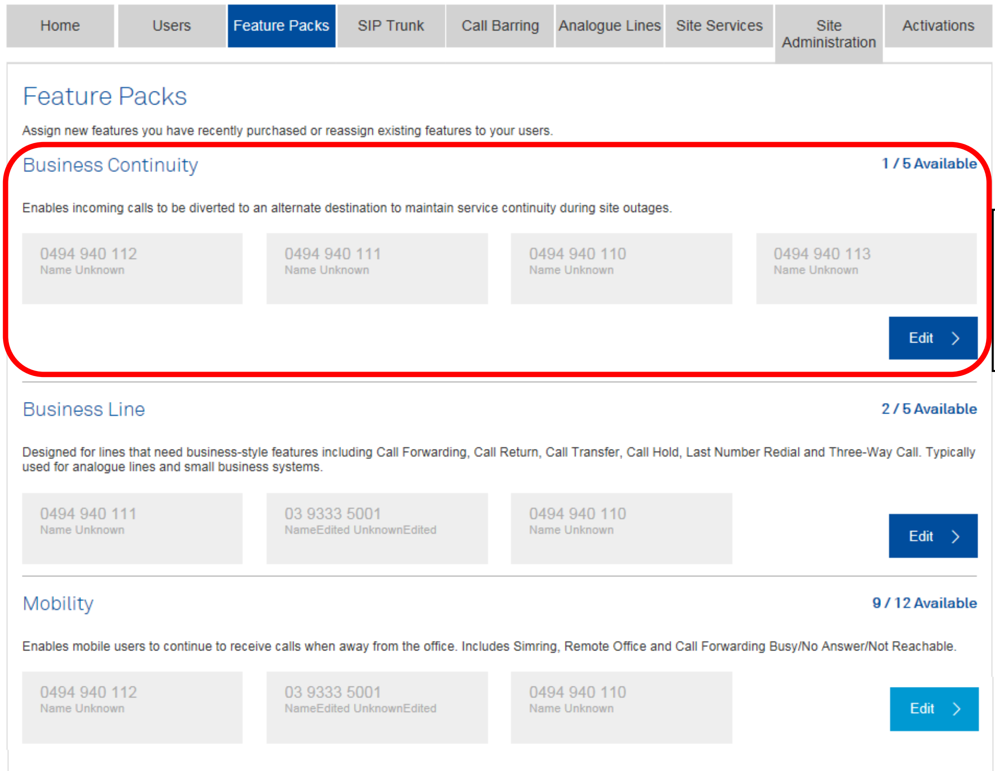
**By feature pack**

1. Select the **Feature Packs** tab.
2. Select the Edit button against the particular feature pack.



### 3.5.2. Making changes to feature packs

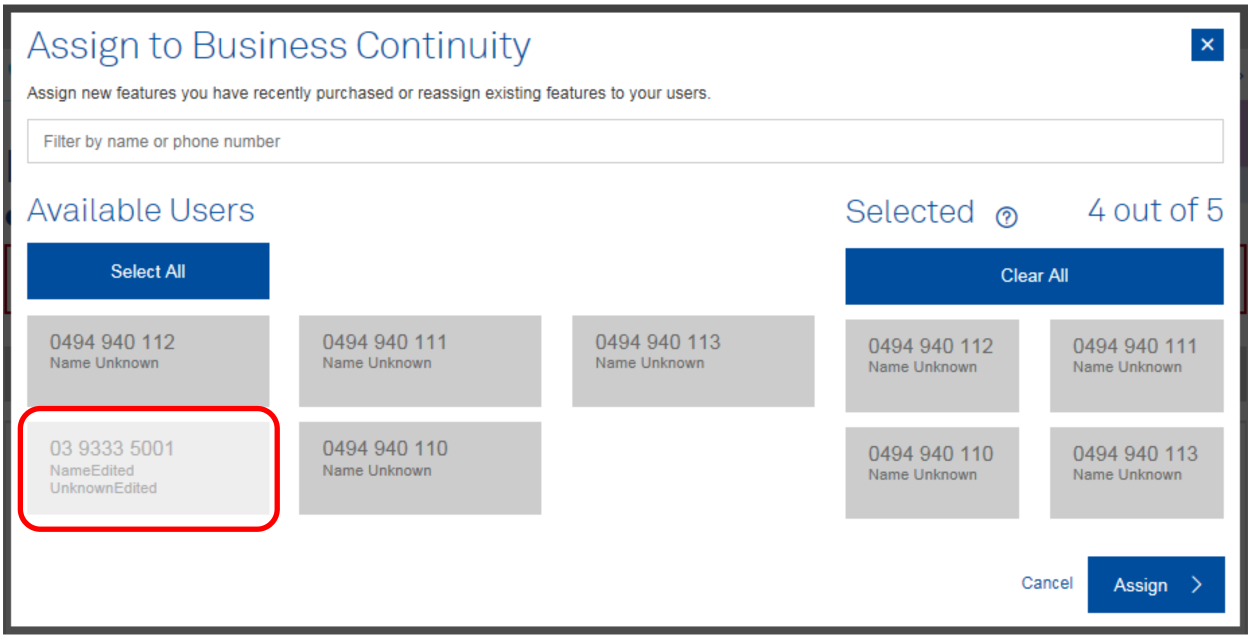
Clicking the Edit button against the particular feature opens the pop-up window, where you can assign/unnassign users.



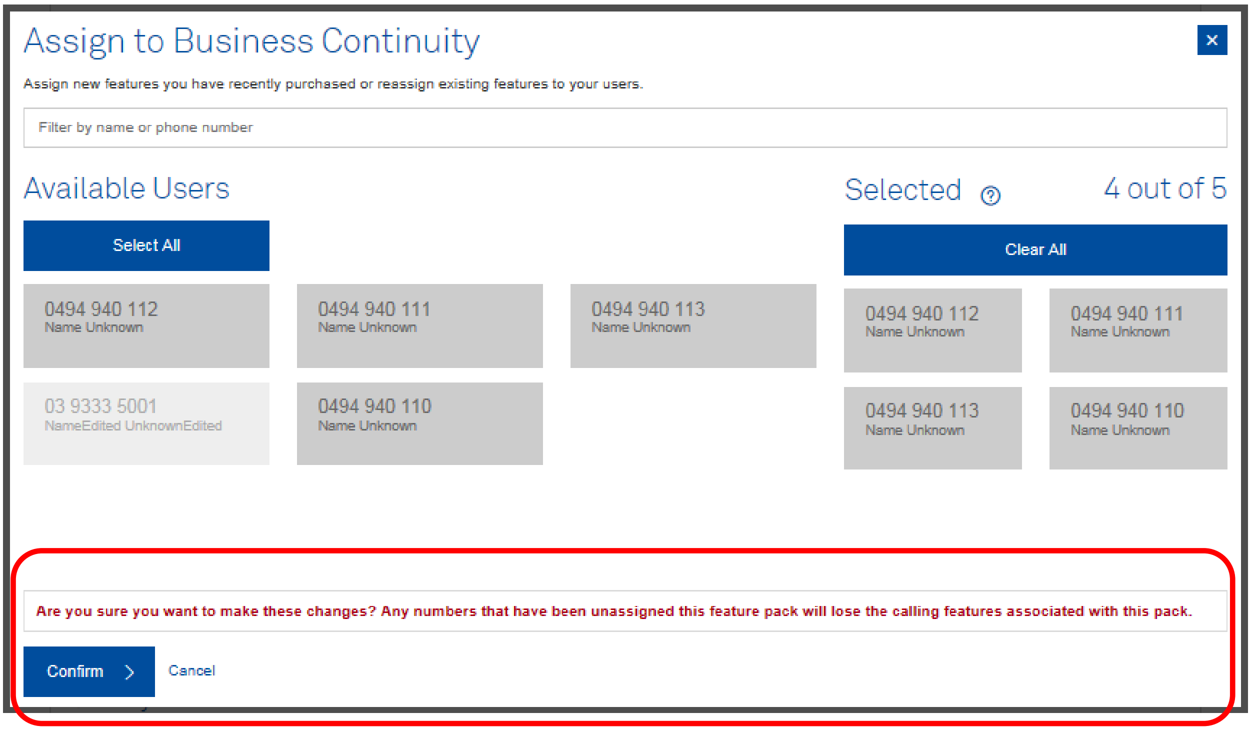

Click the Edit button to make changes to these assignments.

On the left are the available numbers/users and to the right are the numbers/users already assigned. The lighter-shaded box denotes the unassigned user.

To assign or unassign, click on the number/user and it will toggle them.

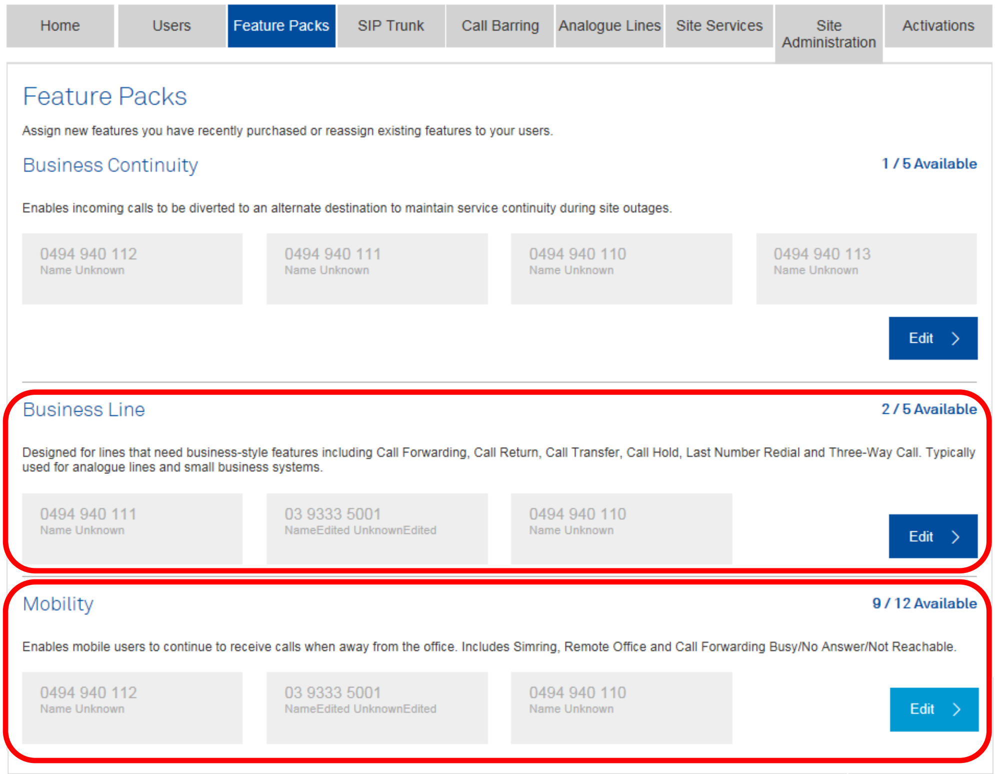



Click Assign to save these changes. You'll be asked to confirm the changes.



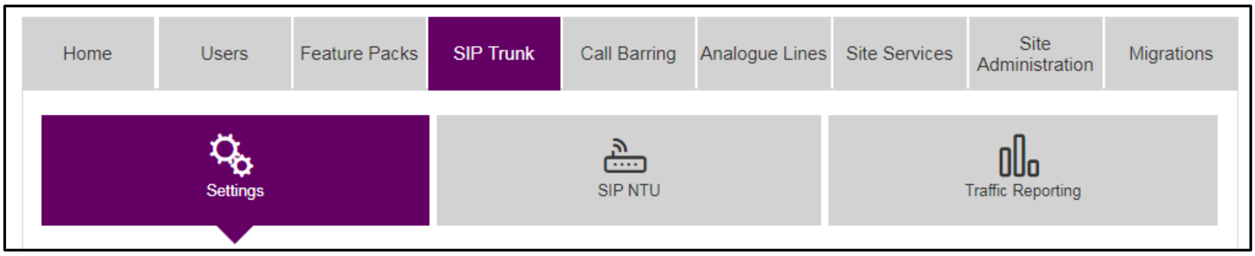
Click Confirm to save.

Use the same method to assign users to Business Line and Mobility Feature Packs.



## 3.6. Set up/configure SIP trunks

Here you can set up and control how your calls arrive at and are made from your business. It is where you set up your SIP NTU device and obtain call traffic reports.




The **Settings** tab contains fields for:

- Trunk partitions (including maximum concurrent calls)
- Calling line ID source
- Business continuity
- Trunk incoming and outgoing call limits

### 3.6.1. Trunk partitions

Trunk partitions allow for phone numbers to be separated out and treated differently to other phone numbers. There are three main reasons for doing this.

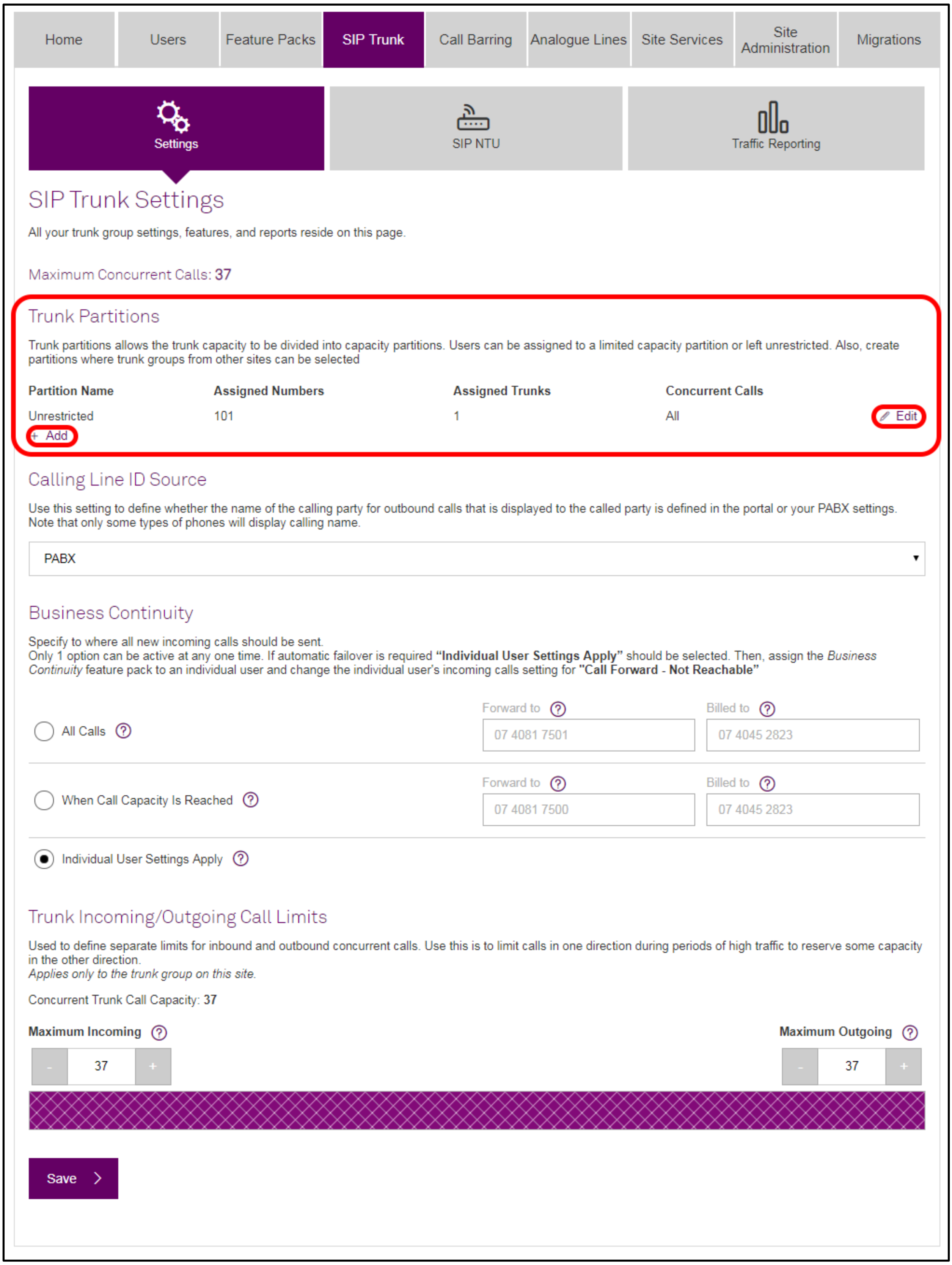
1. Reducing calling capacity available to a group of phone numbers, which reserves capacity for the phone numbers in the unrestricted partition.
2. Giving company departments access to a dedicated capacity partition, so that one department does not use the other department's capacity.
3. Distributing calls to other sites, which may be required where:
    - calls to an advertised number are to be handled at multiple sites
    - staff have temporarily moved to another site and want their number to follow them
    - calls are to be distributed to multiple sites to protect against site failure events (aka Enterprise Trunking)

Delivering phone calls to another site requires that the PBX at the other site has been configured to handle calls to these numbers, as the calls retain the original called number.

The multi-trunk group functionality has no impact on the capacity of the underlying trunk groups hence there is no cost impact.

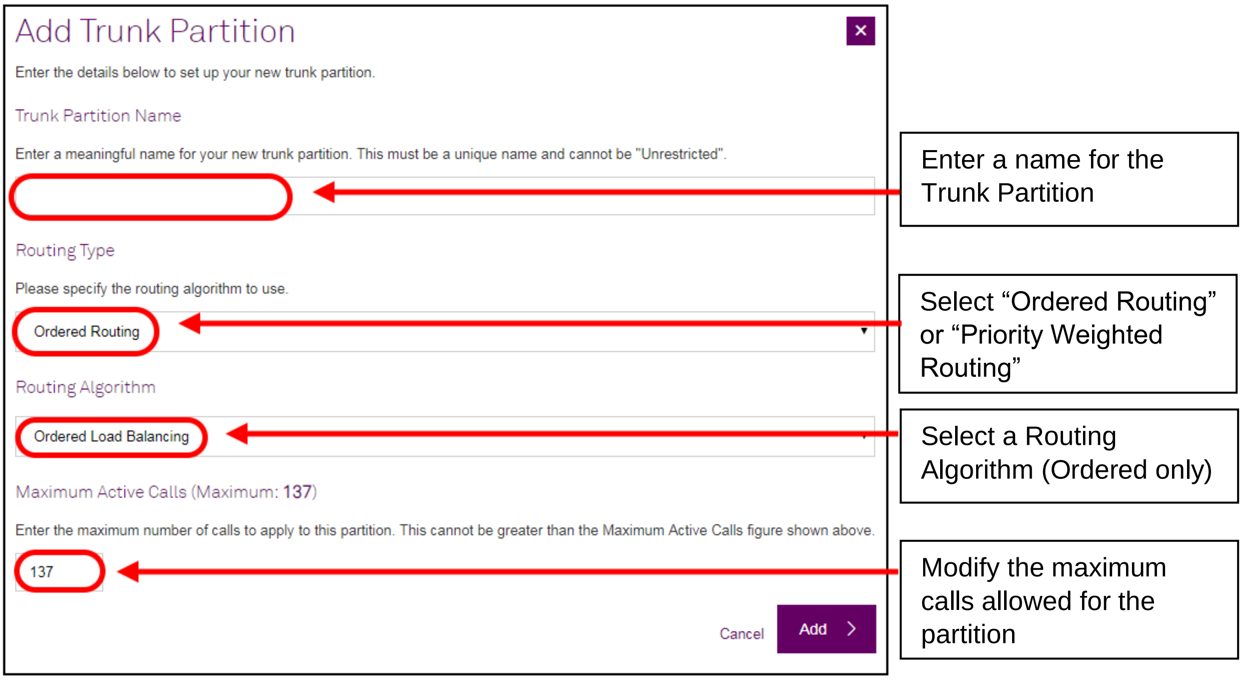
**The Unrestricted partition:** This is the base or default partition that is in place at initial provisioning of the site. Initially it has all phone numbers assigned, has the site trunk group assigned and has unrestricted capacity. The only change that can be made to this partition is reassigning phone numbers to other partitions. If other partitions are deleted, the assigned phone numbers will be automatically moved back to the Unrestricted partition.

Any new phone numbers added to a site will be added to the unrestricted partition.




Highlighted above is the trunk partition section of the web page. Click the Edit icon to make changes to an existing trunk partition or click + Add to create a new one.

The following screen appears when adding a new trunk partition.




**Trunk Partition Name:** At creation time a name is required. Choose something meaningful. Maximum number of characters is 15.

**Routing Type:** Determines how inbound calls are offered to the assigned trunk groups. There are two policy types – Ordered and Priority Weighted. Once this is set for a partition, it is not possible to convert the partition from Ordered to Weighted or from Weighted to Ordered. However, it is possible to change options within a policy type, e.g. change from Most Idle to Least Idle. Policies behave the same for trunk groups that have consumed their capacity or become unreachable due to site failure.

**Ordered Routing:**

**Load Balancing** – Calls are shared across all assigned trunk groups evenly.

**Overflow** – When the first trunk group in the list is at capacity, the next trunk group with idle capacity is selected to deliver the call.

**Most Idle** – The trunk group that has the fewest number of established calls is selected to deliver the call.

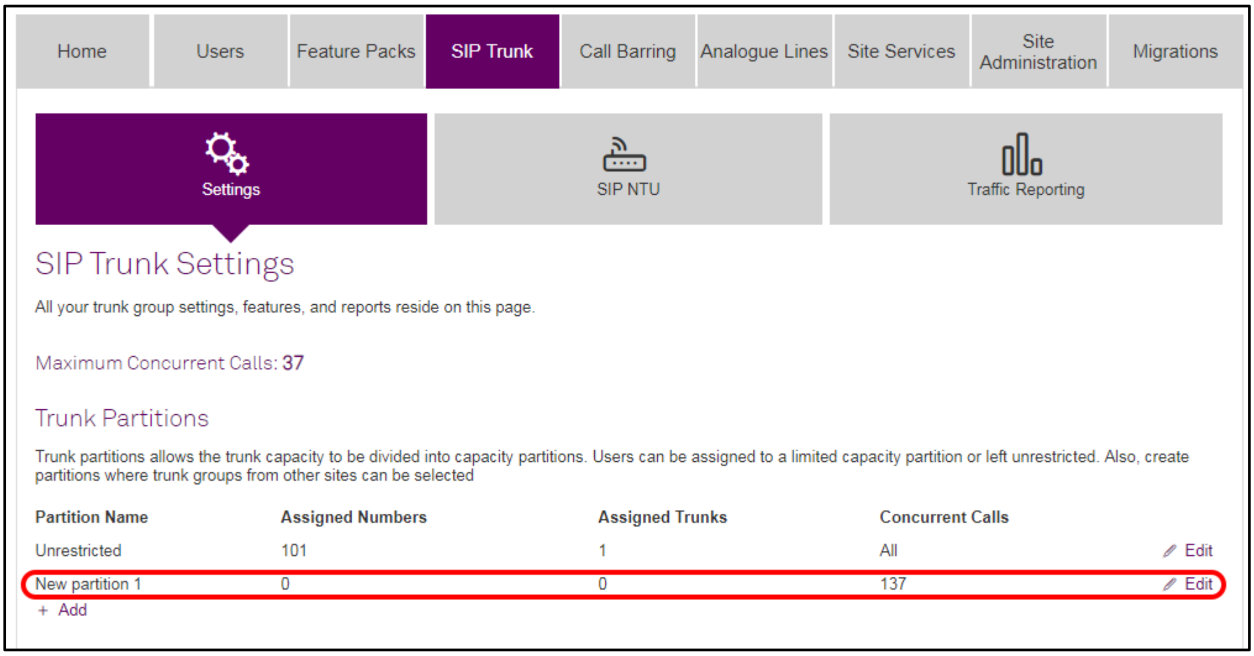
**Least idle** – The trunk group that has the greatest number of established calls and has not reached its capacity is selected to deliver the call.

**Priority Weighted Routing:**

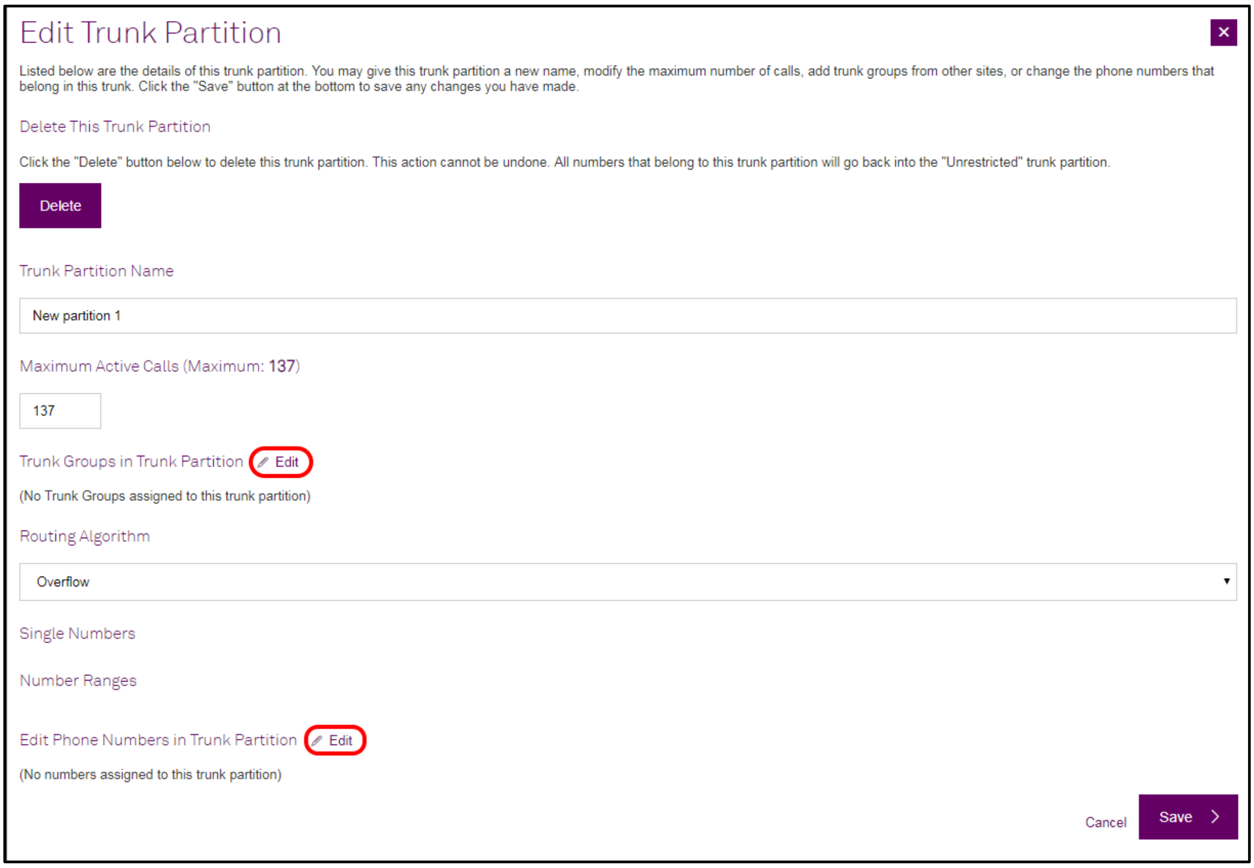
Each trunk group is provisioned with a priority and a weight. Starting with the trunk groups that have the highest priority (the lowest number, between 1 and 10), a trunk group at that priority is selected according to a weighted random pick (from 1 to 100), if more than one trunk group shares the same priority. A trunk group that has reached its capacity cannot be selected.

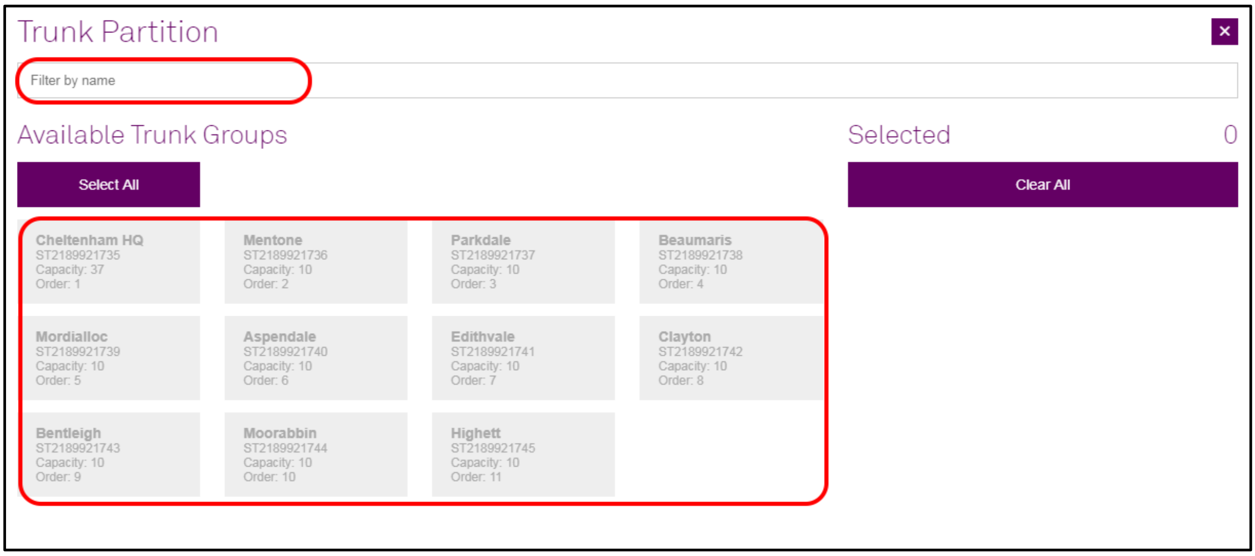
If multiple trunk groups have the same priority, then the weights dictate how many calls each get. For example, if trunk group 1 has a weight of 17 and trunk group 2 has a weight of 26, then, if 43 calls are offered, on average, 17 calls will go to trunk group 1 and 26 calls will go to trunk group 2.

**Maximum Active Calls:** Calls to and from the assigned phone numbers are restricted to the concurrent calling capacity of the trunk partition. Minimum capacity is 1. Maximum capacity is the sum of the capacity of all trunk groups purchased for all sites. Note that the maximum number of concurrent calls at any one time is dictated by the capacity of the underlying trunk groups. Trunk partitions can’t increase the purchased capacity, but they do allow control of where calls go, particularly useful in failure or overflow situations. There is no benefit in assigning more capacity to the partition than capacity of the assigned trunk groups.

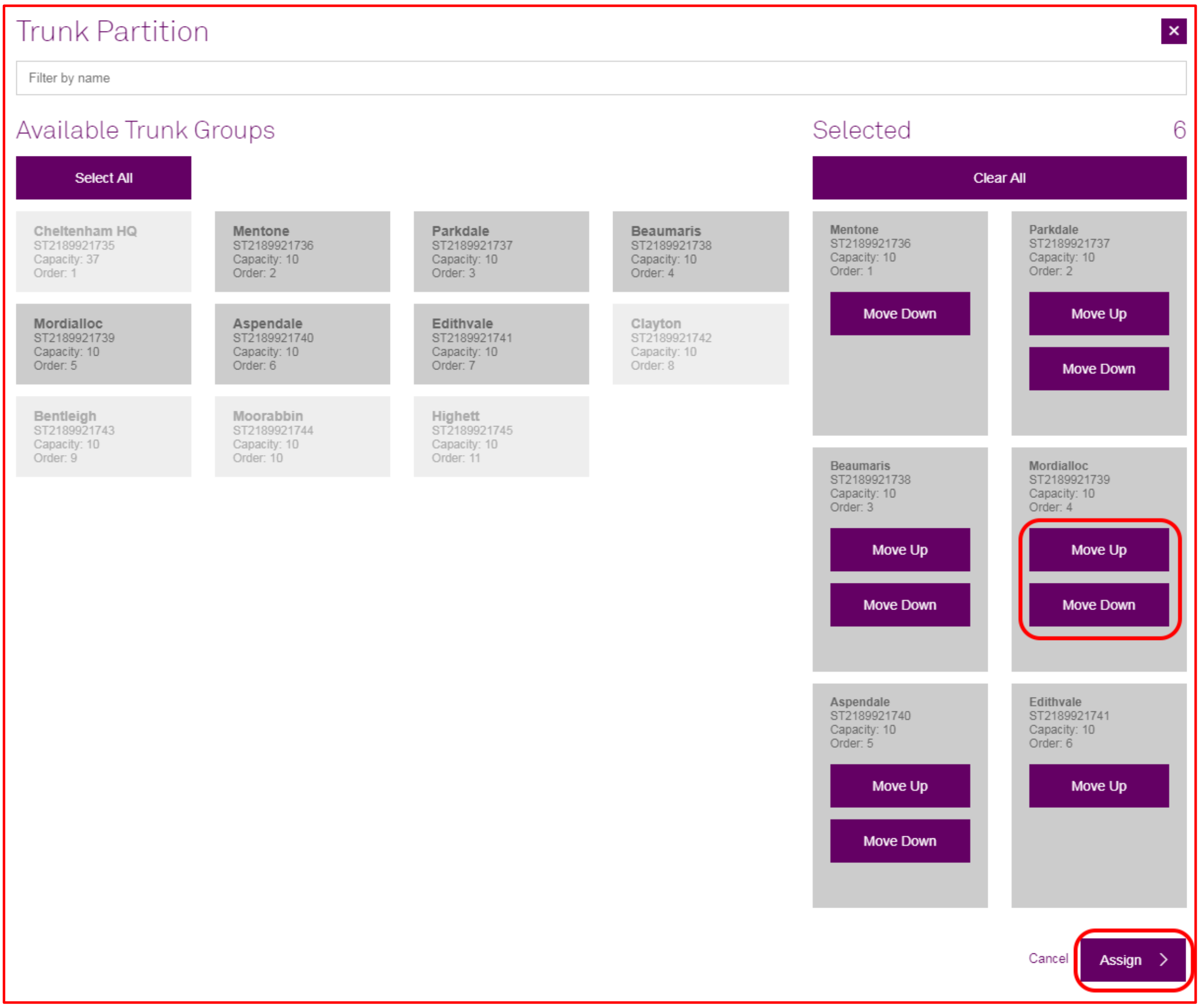



A newly added trunk partition will appear as per the screenshot above. In this case, the name given was “New partition 1”. At this point, there are no assigned numbers or trunk groups. To make these assignments click on “Edit”, which brings up the screen below. Further clicking on “Edit” allows for the trunk group and number assignments to be made.



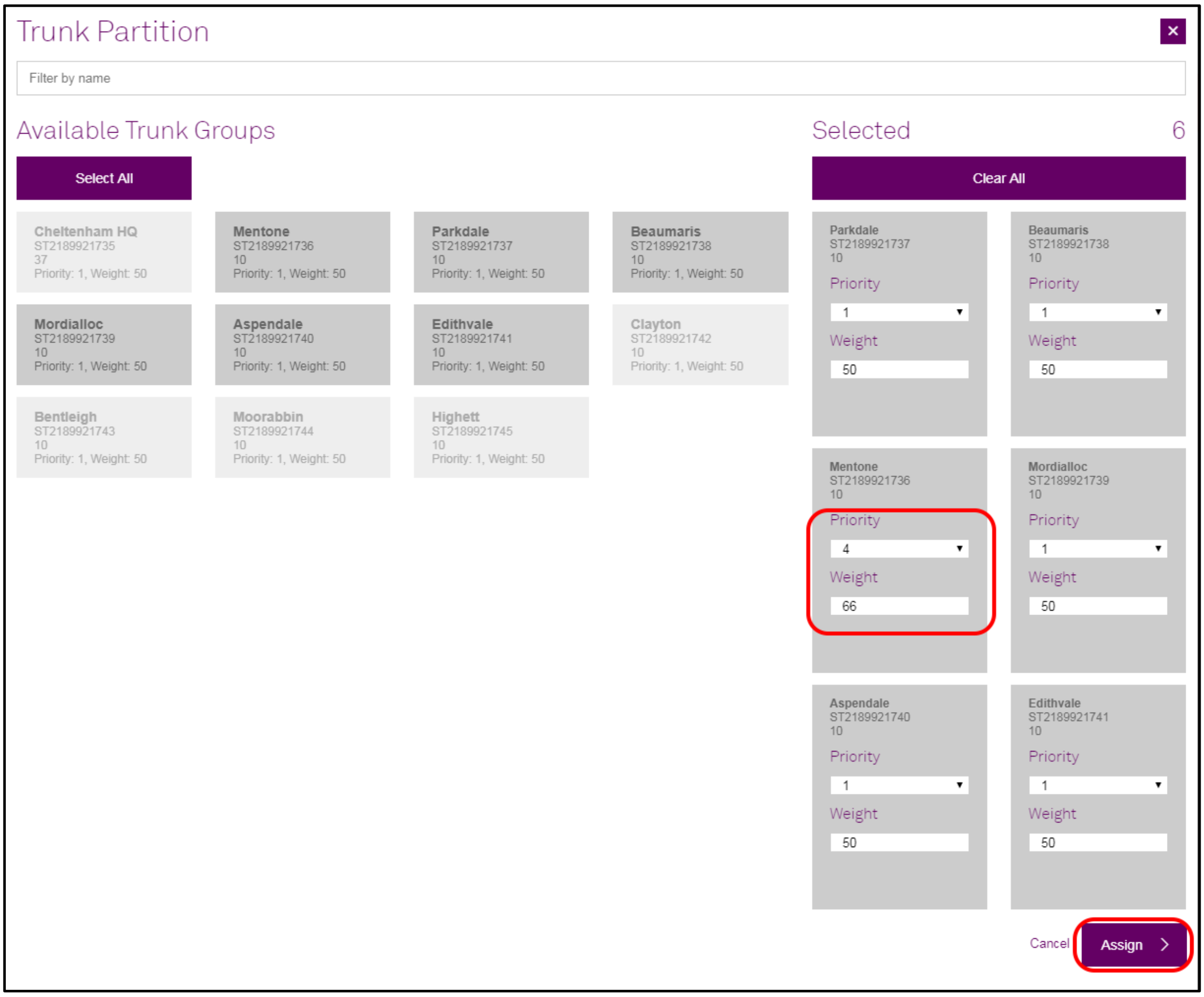



The screen above appears when editing assigned trunk groups. All trunk groups at all sites appear. Select the desired trunk groups (up to 10) with the mouse pointer or search on site name or site ID by typing in the filter field. Selected trunk groups will appear in the "Selected" section on the right-hand side as per the screen below.

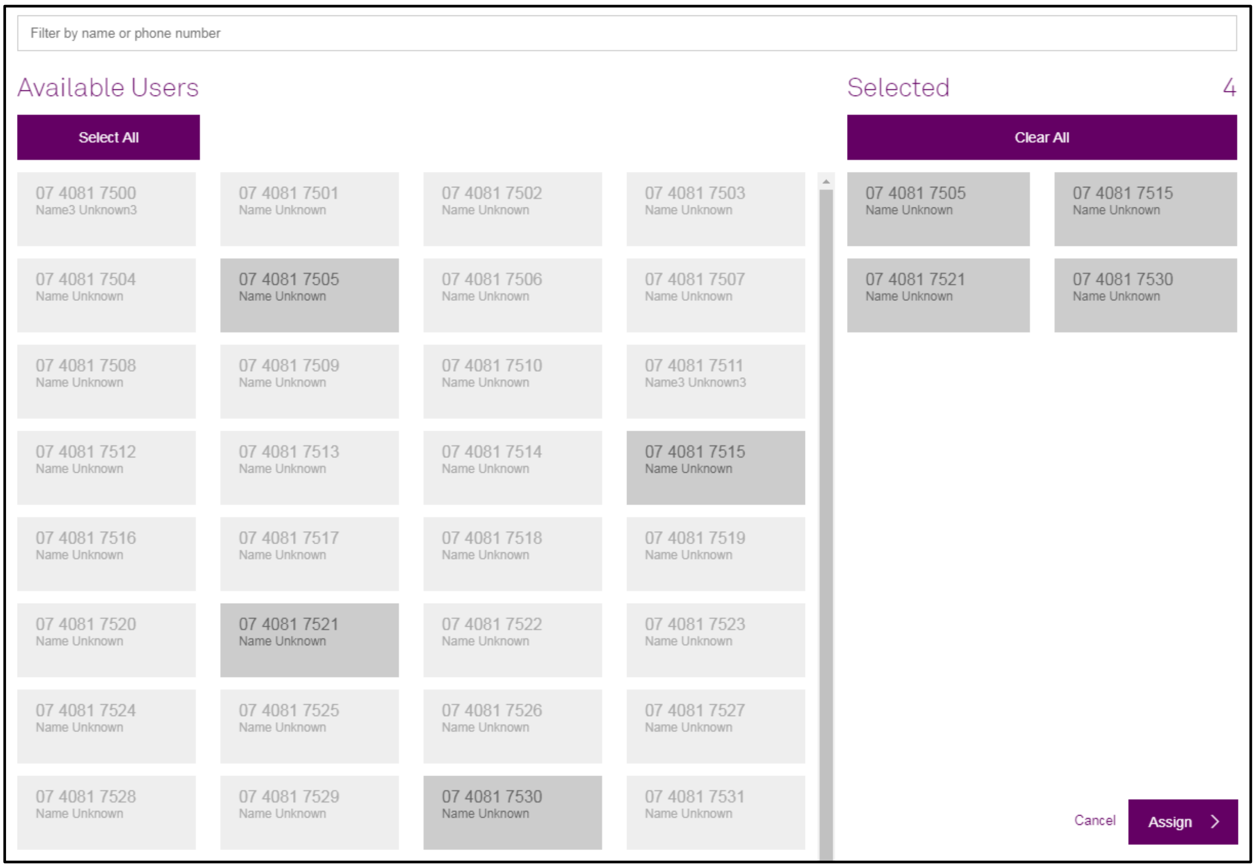


If Overflow is the chosen Routing Algorithm, then the order is important. To change the order, use the “Move Up” and “Move Down” buttons. Click “Assign” to make the change. Order isn’t important for Load balancing, Most Idle and Least Idle.

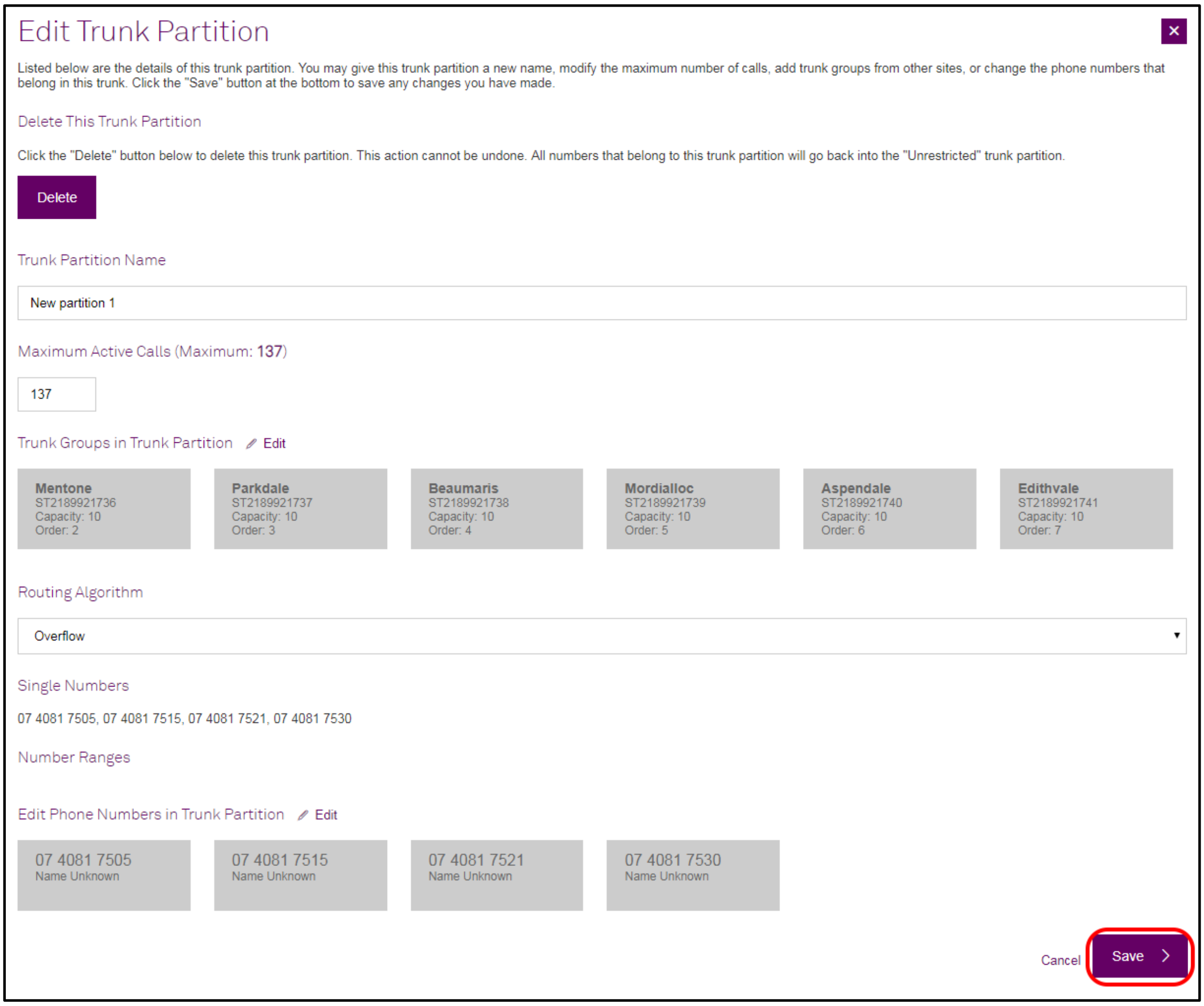
If “Priority Weighted Routing” was chosen, then the screen below appears when assigning trunk groups. Each trunk group has an individual setting for Priority and Weight. Edit each value as needed. Priority can be any value from 1 to 10, with 1 being the highest priority. Weight can be a value from 1 to 100. Click “Assign” to make the change. Note that the weight is only meaningful when there are 2 or more trunk groups with the same priority.



The screen below appears when editing assigned phone numbers. Only phone numbers assigned to the current site appear. Select the desired phone numbers with the mouse pointer or search on phone number or partial phone number by typing in the filter field or select all. Selected phone numbers will appear in the “Selected” section on the right-hand side. Click “Assign” to make the change.

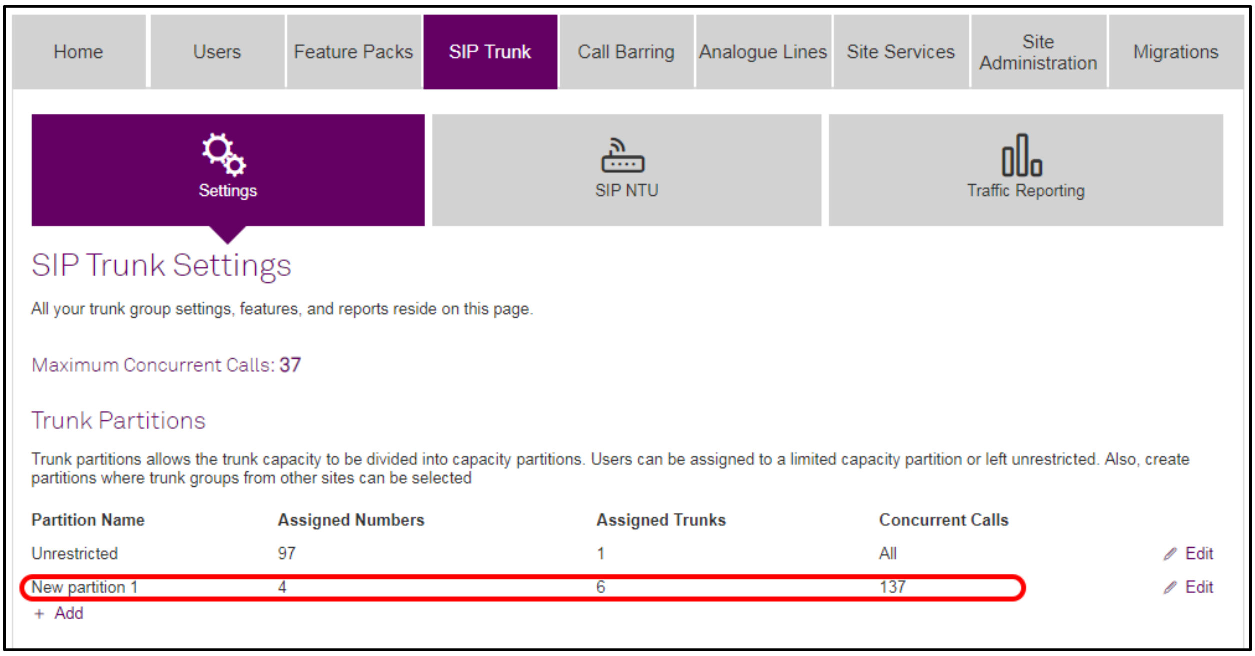


When assignments of trunk groups and phone numbers are complete, the screen below appears. Make a final check before locking it in by selecting “Save”. Phone calls will now follow this new behaviour.



Once all assignments are complete and locked in, the screen below shows the result. Quantities of assigned numbers, assigned trunk groups and concurrent calls are listed. If changes are required, then editing is possible.

Deleting the partition will re-assign its phone numbers to the Unrestricted partition.




Some useful tips:

**Limitations of SIP NTU and State:** At provisioning time, a SIP NTU is assigned to an Australian state, in line with the phone numbers in the sales order. This assignment tells the SIP NTU what STD code (e.g. 02 for NSW, 03 for VIC) to prepend when the full national number is not present in the calling line id on an outbound call. For this reason, it is important, where phone numbers from multiple states are in play, to always ensure that PBX's are configured to present the STD prefix in the calling line id of outbound calls (i.e., full 10-digit numbering).

**Interaction with other features:** It is recommended that employing trunk partitions to distribute calls to multiple trunk groups not be combined with other features such as Business Continuity and other calling features, as the results may not be as expected.

**Deleting trunk capacity:** If proposing to delete trunk capacity and that capacity is configured for use by a trunk partition, then de-allocate that capacity before placing the order. Failure to do so will cause the order to fail. For example, consider a scenario where both Site A and Site B have a trunk group with 200 concurrent call capacity and a trunk partition is configured with 400 concurrent call capacity, with both Site A and Site B trunk groups assigned. If Site B trunk group is to be reduced to 100 concurrent call capacity, then ensure the trunk partition is reduced to 300 concurrent call capacity before the order is placed.

### 3.6.2. Calling line ID source

Use this setting to define whether the displayed name of the outbound *calling* party (which is presented to the *called* party) is:

- defined in the Customer Management Portal
- configured in the PBX settings.

**Note:** Only some types of phones will display calling name.

### 3.6.3. Business continuity

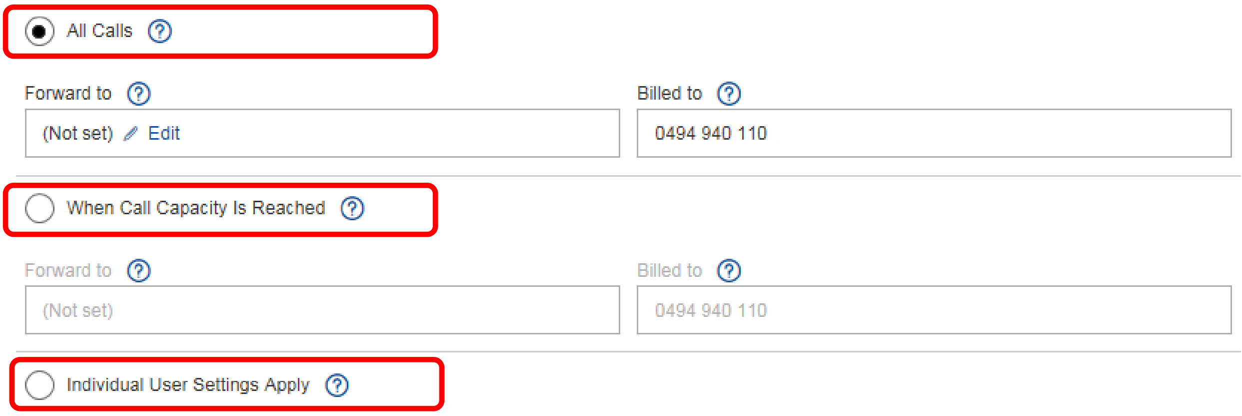
Specify the number where new incoming calls should be sent, for any of the following situations:

**All calls** – All new incoming calls will be forwarded to the specified (forward to) number.

**When call capacity is reached** – When your trunk group capacity is reached, all new incoming calls will be forwarded to the specified (forward to) number.

**Individual user setting apply** – Use the individual user level settings as required. See section 4.3 Incoming call features – activate/deactivate/configure.

**Note:** Only one option can be active at any one time.

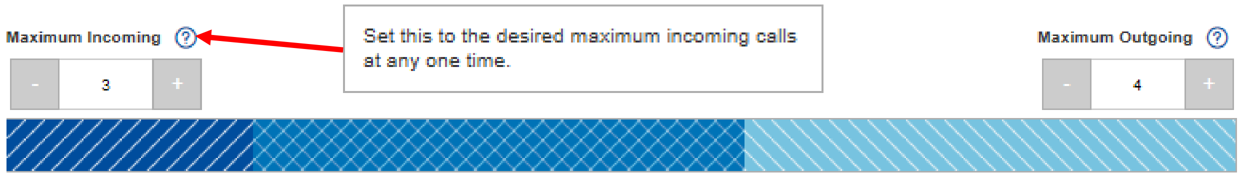


### 3.6.4. Trunk incoming/outgoing call limits

Used to define separate limits for inbound and outbound concurrent calls. Use this to limit calls in one direction during periods of high traffic – this reserves some capacity in the other direction.

In the example below, the limit of three incoming calls means that at least two outgoing services will always be available (light blue shading), while the limit of four outgoing calls means that there will always be one service available for incoming (dark blue shading). Within these limitations **only five calls** can be in progress at any one time.

Concurrent trunk call capacity: **5**




After making your required changes click Save .

## 3.7. Change CPE device/retrieve SIP NTU credentials

**IMPORTANT NOTE: PLEASE READ THE FOLLOWING BEFORE CHANGING THE SIP NTU**

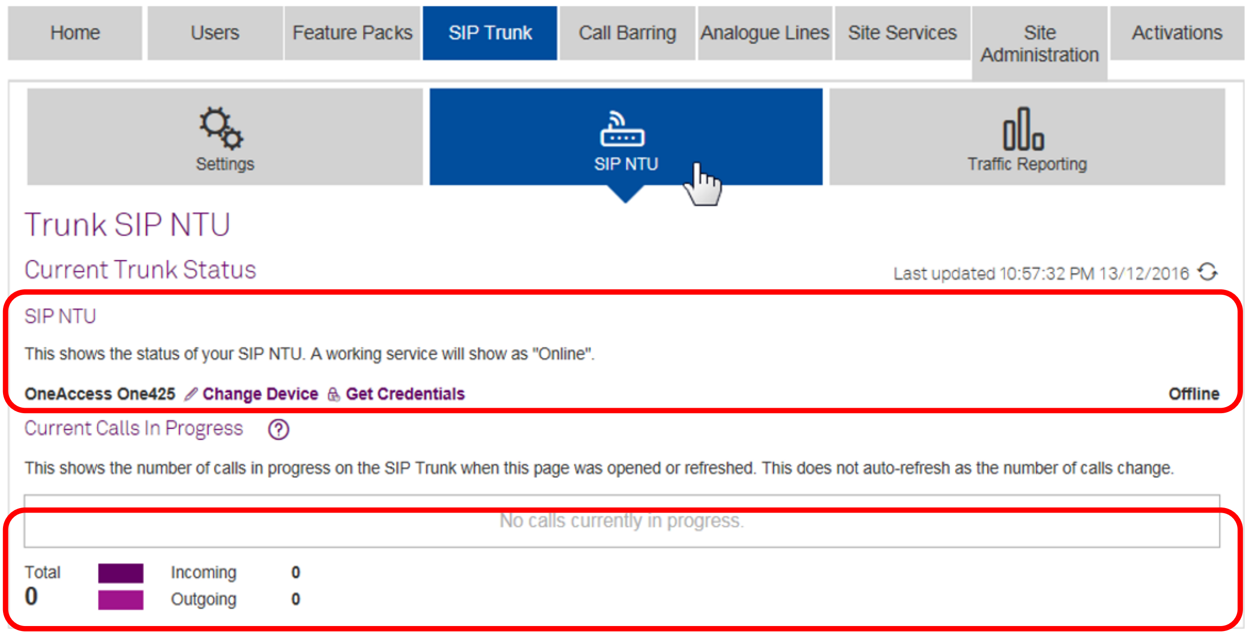
Changing the SIP NTU will mean that your **services will be interrupted** until the new device is fully configured to the Business SIP server (northbound) and the PBX (southbound).

The installation/configuration will need to be **performed by an appropriately skilled technician following the particular SIP NTU installation/configuration guides**.

Should you wish to **re-use your original device**, you will need to repeat the installation and configuration using the appropriate resources, i.e. skilled technicians and guides, as stated above.

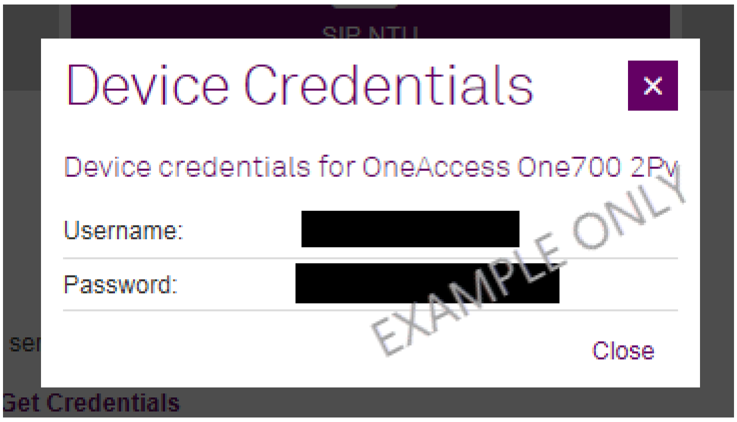
To change the SIP NTU device or to retrieve the associated credentials, go to the SIP Trunk tab then select SIP NTU SIP NTU as shown. The initial display shows the existing device details and **current calls**; which is important as changing the device will impact any calls in progress. Check the date and time stamp on the right and click the refresh icon as required to ensure that you have the latest information displayed.

The SIP NTU credentials can be accessed by clicking **Get Credentials** as shown in the screen below:

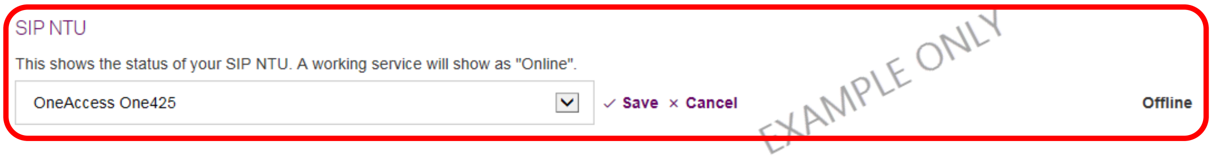


Clicking **Get Credentials** opens a pop-up window presenting the username and password.

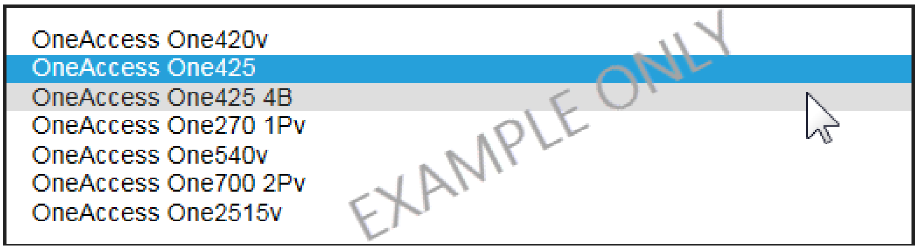
**Note:** Keep these details strictly confidential as the integrity of your device depends on this information.




To change this device, select Change Device to open the dropdown list and the ✓ Save / × Cancel options.



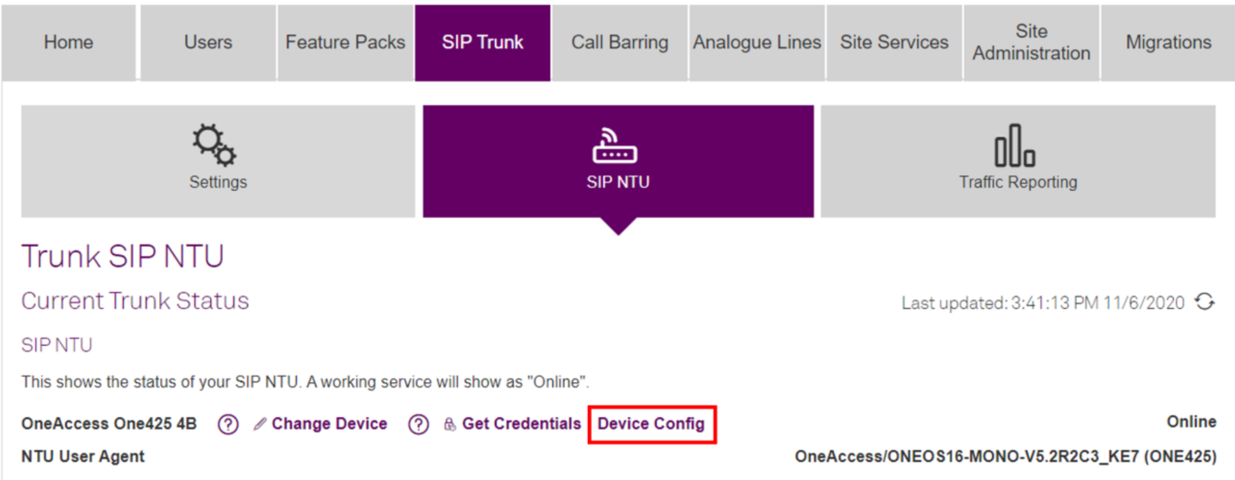

The dropdown list presents the selection of available SIP NTU devices with the current device highlighted in blue and the new selection in grey (when you hover your mouse over it):



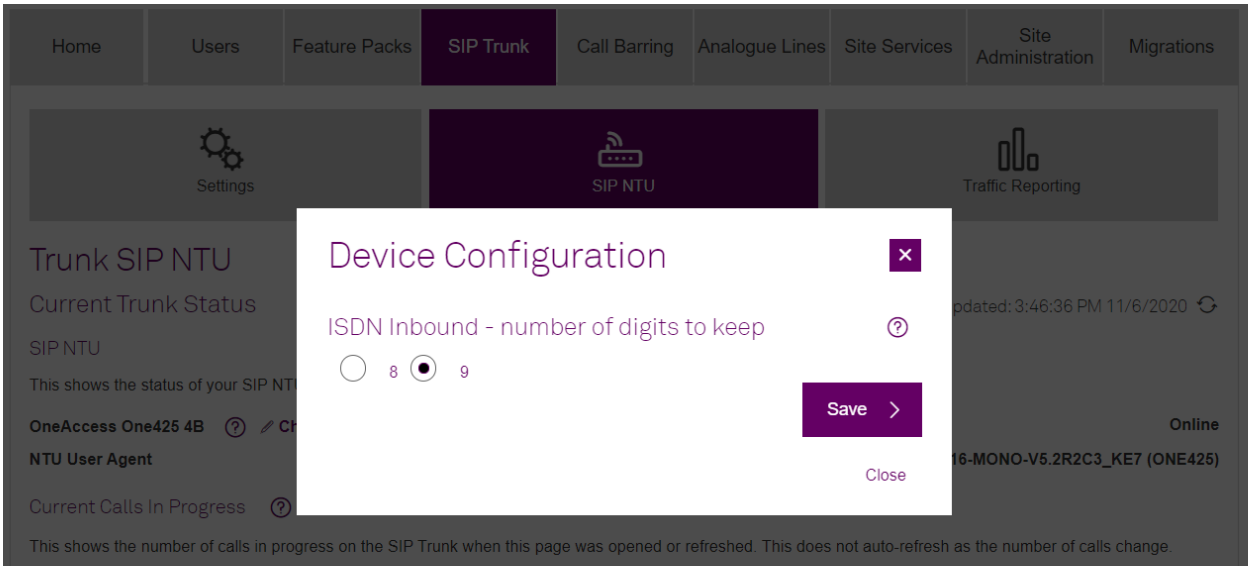
## 3.8. ISDN inbound – number of digits to keep

This is a SIP NTU setting that determines how many dialled digits are presented to the connected PABX on inbound calls, where the ISDN interface is being employed. The setting is normally applied on the SIP NTU, at the time of installation, using the web GUI of the device. It can also be applied via the Business SIP Customer Portal in case it was missed at installation time.

Navigate to the SIP NTU tab and click “Device Config”.




On the “Device Configuration” pop-up, select 8 or 9 digits to keep, and click “Save”.

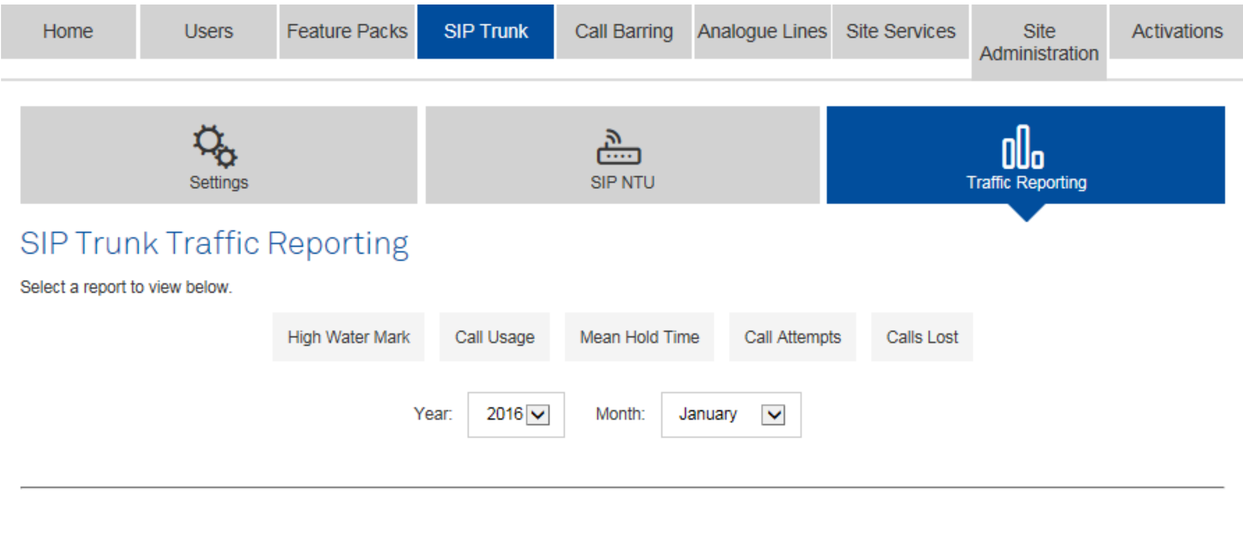



Power-cycle the SIP NTU so that the setting can be transferred to the device.

If the setting on the device has already been made, and the system is working satisfactorily, there is no need to manipulate the setting on the Business SIP Customer Portal.

## 3.9. Traffic reporting

Call traffic reports are available to administrators on the **SIP Trunk** tab.



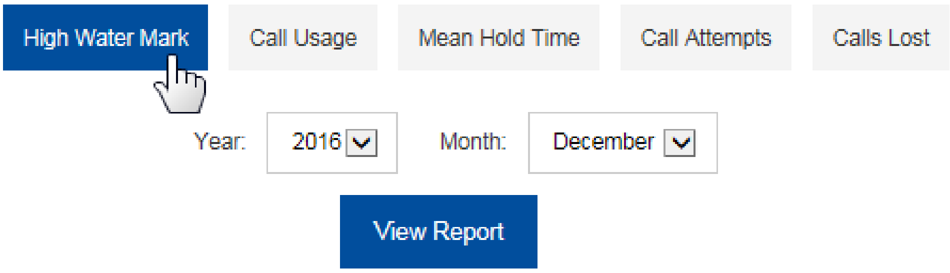

Descriptions of the five report types is provided below:

1. **High watermark** – the highest number of calls in progress seen at any point in the hour. There are three measures: **Incoming, Outgoing** and **Both ways**.
2. **Call usage** – the average number of calls in progress over the trunk group over the hour. This is achieved by taking regular samples of calls in progress and averaging the results. There are three measures: **Incoming, Outgoing** and **Total**.
3. **Mean hold time** – the average length of calls on the trunk group over the hour. Unsuccessful calls are included in the measure which means that an increase in unsuccessful calls drags down the mean holding time. The three measures are **Incoming, Outgoing** and **Both ways**.
4. **Call attempts** – the count of new calls over the trunk group over the hour. Also shown is calls blocked for any reason. The two measures are **Incoming and Outgoing**.
5. **Calls lost** – the three measures are:
    a. **Capacity exceeded** – count of incoming and outgoing calls blocked due to the capacity of the trunk group being exceeded
    b. **Incoming failure reason: Unreachable** – count of incoming calls failed due to the SIP NTU being unreachable
    c. **Incoming failure reason: Failure** – count of incoming calls failed due to an unspecified failure condition

The following shows the progression through the steps, which is the same for all reports.

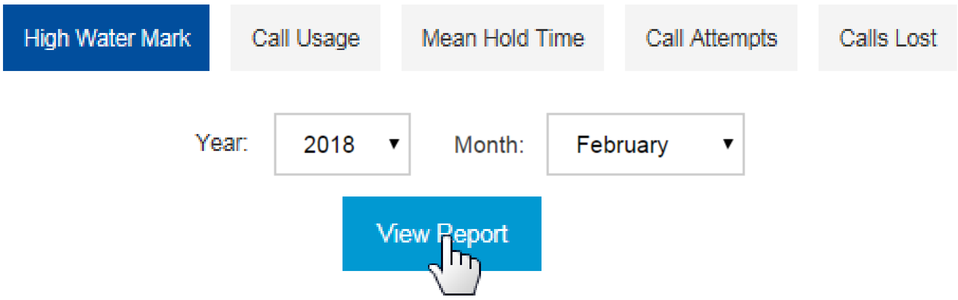
**Example:** High watermark report

1. Select the report type

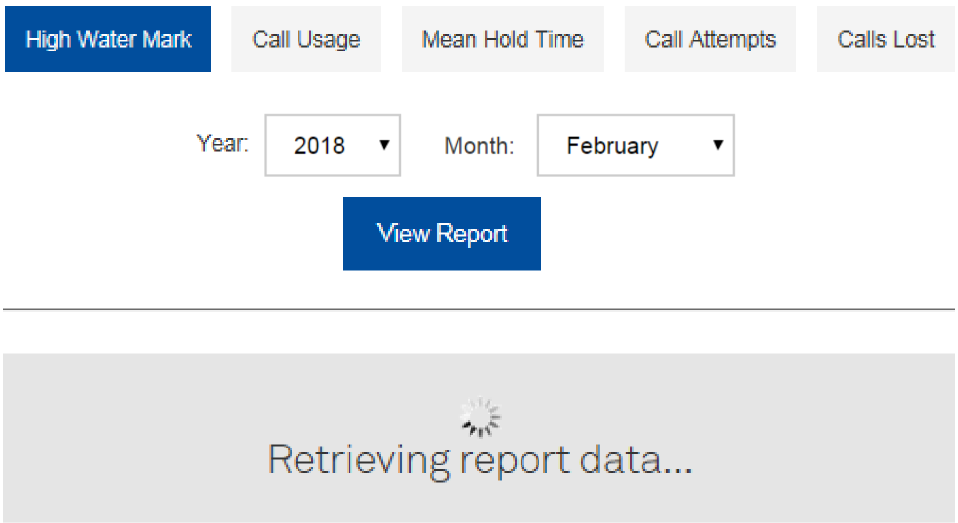



2. Use the dropdowns to select


3. Click **View Report**




4. The report data will be retrieved.




Follow the above steps for the other report types.

Below is a sample of the High Water Mark report showing the graph and tabular representation as well as the button to download the data in CSV format.

**Note:** The sample cannot show all of the tabular data, which is reported hourly.

Settings | SIP NTU | Traffic Reporting

## SIP Trunk Traffic Reporting

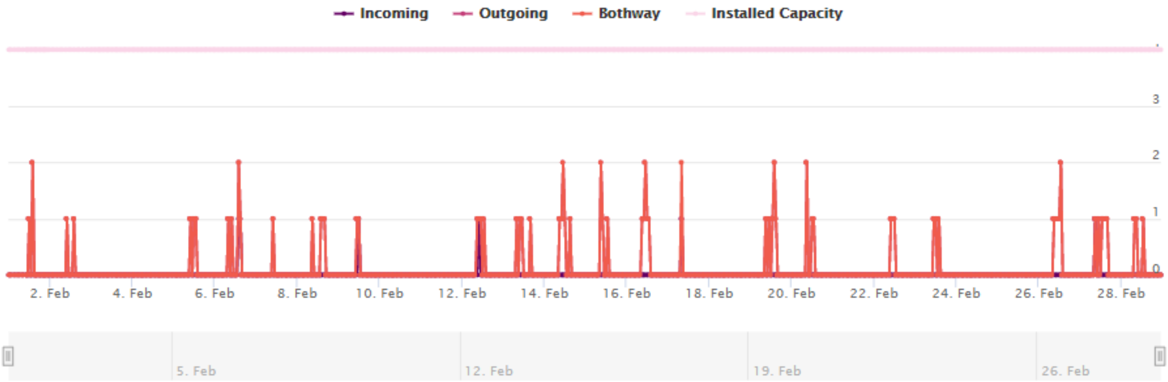
Select a report to view below.

High Water Mark | Call Usage | Mean Hold Time | Call Attempts | Calls Lost

Year: 2018 ▼ Month: February ▼

View Report

High Water Mark (calls)

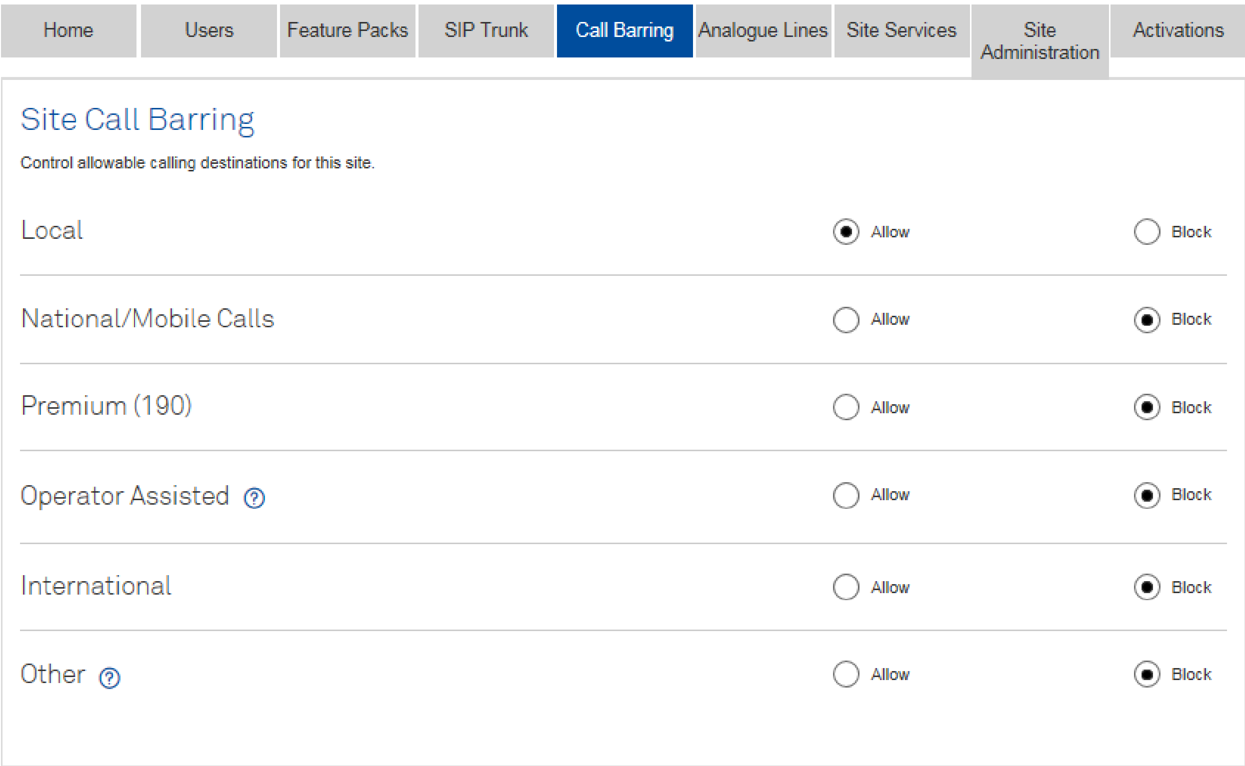



Download as CSV

| Date (Local Time) | Incoming | Outgoing | Bothway | Installed Capacity |
|---|---|---|---|---|
| 1 Feb, 0:00 AEDT | 0 | 0 | 0 | 4 |
| 1 Feb, 1:00 AEDT | 0 | 0 | 0 | 4 |
| 1 Feb, 2:00 AEDT | 0 | 0 | 0 | 4 |
| 1 Feb, 3:00 AEDT | 0 | 0 | 0 | 4 |
| 1 Feb, 4:00 AEDT | 0 | 0 | 0 | 4 |
| 1 Feb, 5:00 AEDT | 0 | 0 | 0 | 4 |
| 1 Feb, 6:00 AEDT | 0 | 0 | 0 | 4 |
| 1 Feb, 7:00 AEDT | 0 | 0 | 0 | 4 |
| 1 Feb, 8:00 AEDT | 0 | 0 | 0 | 4 |
| 1 Feb, 9:00 AEDT | 0 | 0 | 0 | 4 |
| 1 Feb, 10:00 AEDT | 0 | 0 | 0 | 4 |
| 1 Feb, 11:00 AEDT | 0 | 0 | 0 | 4 |
| 1 Feb, 12:00 AEDT | 0 | 1 | 1 | 4 |
| 1 Feb, 13:00 AEDT | 0 | 0 | 0 | 4 |
| 1 Feb, 14:00 AEDT | 0 | 2 | 2 | 4 |
| 1 Feb, 15:00 AEDT | 0 | 0 | 0 | 4 |
| 1 Feb, 16:00 AEDT | 0 | 0 | 0 | 4 |
| 1 Feb, 17:00 AEDT | 0 | 0 | 0 | 4 |
| 1 Feb, 18:00 AEDT | 0 | 0 | 0 | 4 |
| 1 Feb, 19:00 AEDT | 0 | 0 | 0 | 4 |
| 1 Feb, 20:00 AEDT | 0 | 0 | 0 | 4 |
| 1 Feb, 21:00 AEDT | 0 | 0 | 0 | 4 |
| 1 Feb, 22:00 AEDT | 0 | 0 | 0 | 4 |
| 1 Feb, 23:00 AEDT | 0 | 0 | 0 | 4 |
| 2 Feb, 0:00 AEDT | 0 | 0 | 0 | 4 |
| 2 Feb, 1:00 AEDT | 0 | 0 | 0 | 4 |
| 2 Feb, 2:00 AEDT | 0 | 0 | 0 | 4 |
| 2 Feb, 3:00 AEDT | 0 | 0 | 0 | 4 |

## 3.10. Set up site call barring

Call barring is controlled at the site level to provide control over the types of calls that can be made. Use the radio buttons to **Allow** or **Block** for each category.




When a change is made the Save button appears at the bottom of the page.

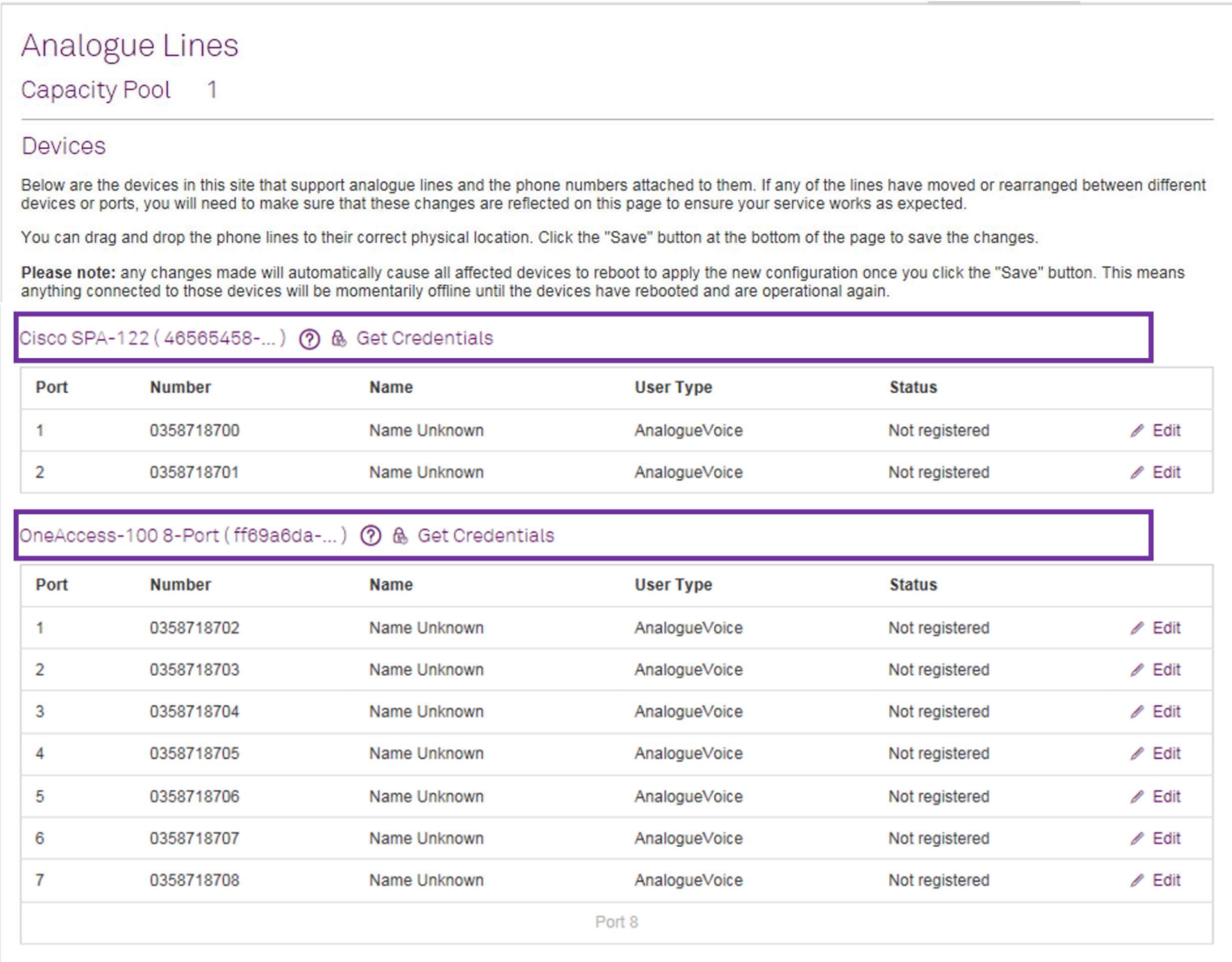
## 3.11. Analogue lines/configuration/get credentials

Analogue lines are mainly used for devices such as fax machines and dial-up modems and are delivered by the use of an IAD (Integrated Analogue Device) that supports analogue lines.

This section describes how the related information is presented and shows two IADs with their respective phone numbers.

The associated device credentials are also accessible from this location.

The Capacity Pool relates to the quantity of Analogue Calling Plans purchased for the site. This should equate to the maximum number of calls you want to have in progress at any one time. In the example only one call can happen at one time, which will not work very well with 9 lines!



Analogue Lines

Capacity Pool 1

Devices

Below are the devices in this site that support analogue lines and the phone numbers attached to them. If any of the lines have moved or rearranged between different devices or ports, you will need to make sure that these changes are reflected on this page to ensure your service works as expected.

You can drag and drop the phone lines to their correct physical location. Click the "Save" button at the bottom of the page to save the changes.

**Please note:** any changes made will automatically cause all affected devices to reboot to apply the new configuration once you click the "Save" button. This means anything connected to those devices will be momentarily offline until the devices have rebooted and are operational again.

Cisco SPA-122 ( 46565458-... ) Get Credentials

| Port | Number | Name | User Type | Status | |
|---|---|---|---|---|---|
| 1 | 0358718700 | Name Unknown | AnalogueVoice | Not registered | Edit |
| 2 | 0358718701 | Name Unknown | AnalogueVoice | Not registered | Edit |

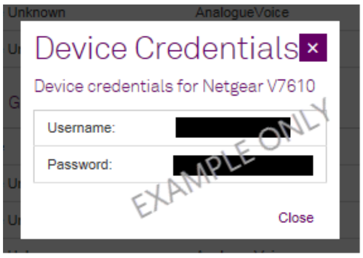
OneAccess-100 8-Port ( ff69a6da-... ) Get Credentials

| Port | Number | Name | User Type | Status | |
|---|---|---|---|---|---|
| 1 | 0358718702 | Name Unknown | AnalogueVoice | Not registered | Edit |
| 2 | 0358718703 | Name Unknown | AnalogueVoice | Not registered | Edit |
| 3 | 0358718704 | Name Unknown | AnalogueVoice | Not registered | Edit |
| 4 | 0358718705 | Name Unknown | AnalogueVoice | Not registered | Edit |
| 5 | 0358718706 | Name Unknown | AnalogueVoice | Not registered | Edit |
| 6 | 0358718707 | Name Unknown | AnalogueVoice | Not registered | Edit |
| 7 | 0358718708 | Name Unknown | AnalogueVoice | Not registered | Edit |
| Port 8 | | | | | |

Clicking the Edit icon presents the **User Details** page, where you can update the user name/password and associated feature packs.

Clicking **Get Credentials** opens a pop-up window presenting the username and password.

**Note:** Keep these details strictly confidential as the integrity of your device depends on this information.



### 3.12. Hunt group services

If your business uses multiple analogue users/agents, the hunt group site service can control how calls are distributed across them. There are two ways calls can be distributed:

• **Regular** – sends incoming calls to the next available user/agent in the hunt group, always starting with the first agent on the list, with optional overflow on 'no answer' to next member.

• **Simultaneous** – sends incoming calls to all users'/agents' numbers at the same time. Once the call has been answered, ringing to other users/agents stops.

You can enable call waiting for an analogue user/agent where, if the user/agent is already on a call, the new call is still offered. This is set up for individual users/agents through the **Incoming Calls** feature under the **Users** tab.

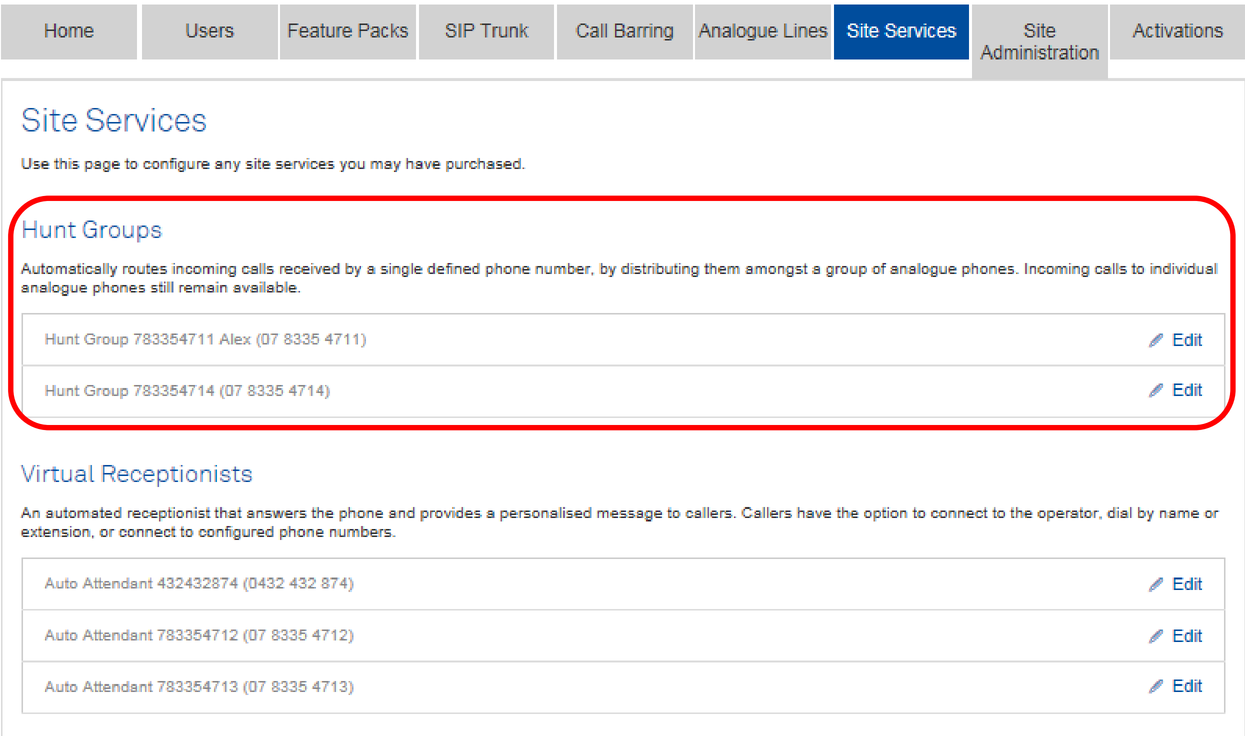
In the event that there is no answer from the user/agent, the following options are available:

• **Skip to the next agent** – the number of rings is configurable

• **Forward calls** – the time and number to forward calls to is configurable.

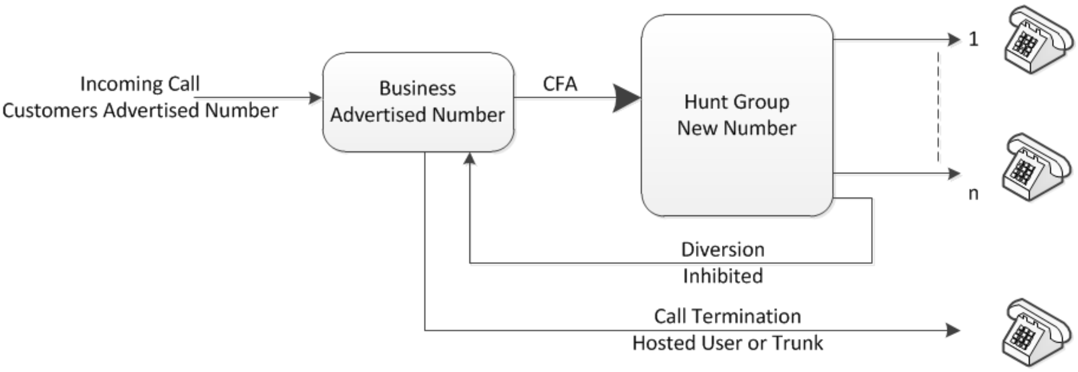
In the event that all users/agents in the hunt group are busy, the call forward busy treatment applies. For this to be active, you need to configure the call forward number.

Forward calls in the above case can be directed to voicemail or any destination phone number (e.g. IP telephony, mobile, PSTN). For voicemail to be active, the voicemail service must be provisioned.

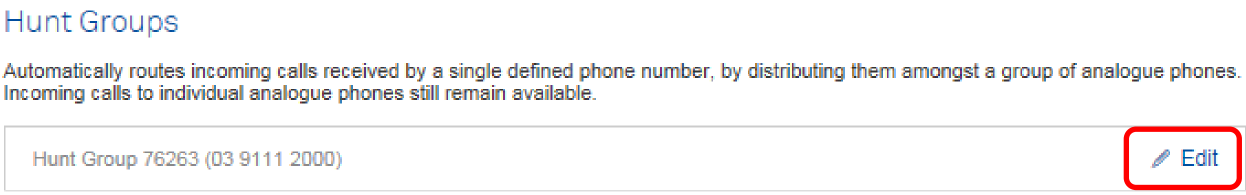


The hunt group is provisioned with its own number and is a simple, flexible solution. Any phone number (e.g. a customer's advertised phone number) can be forwarded to the hunt group and still be a terminating station on the hunt group; this is because all stations in a hunt group have an inherent diversion inhibitor.

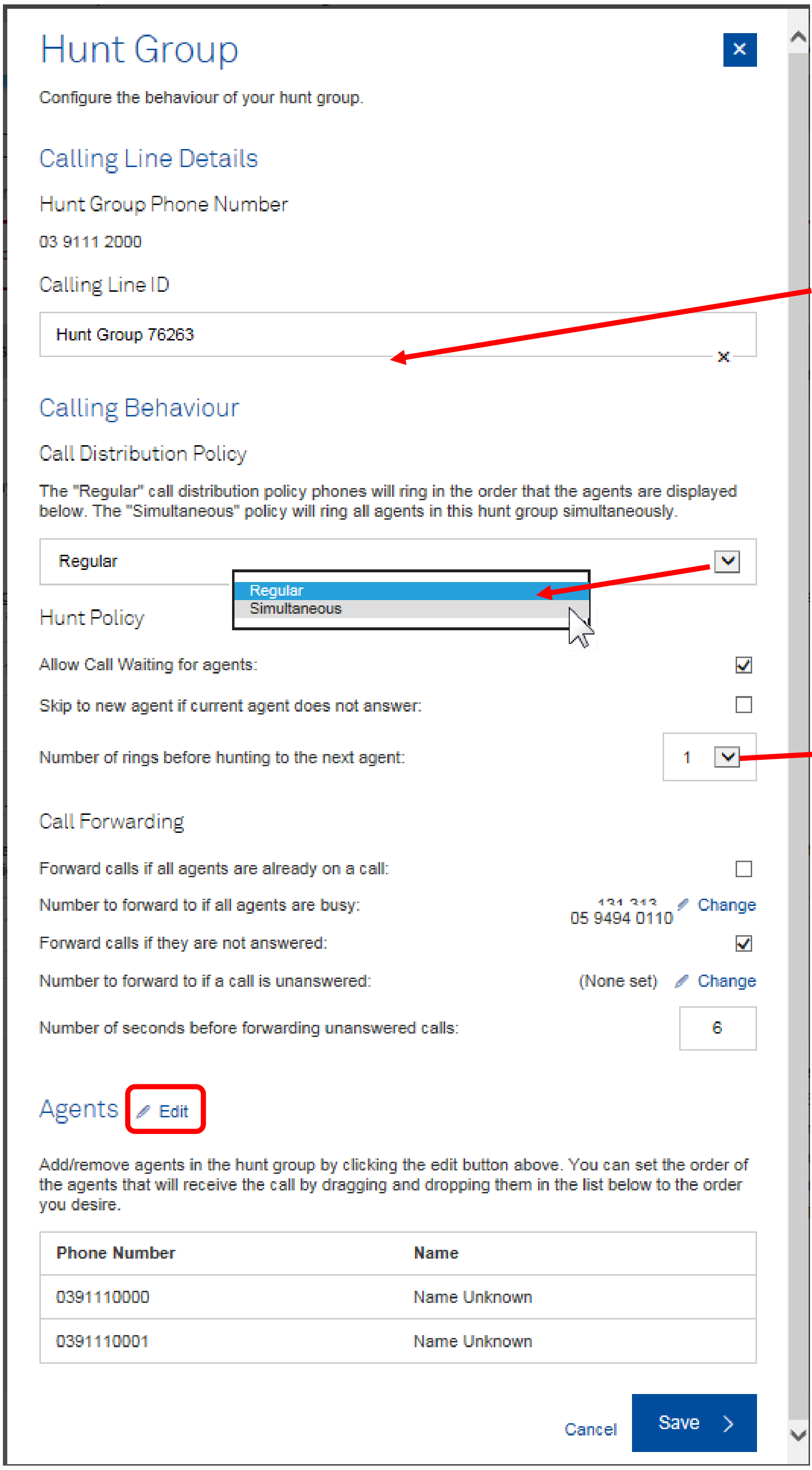
The figure below shows the hunt group functionality as it would be implemented.



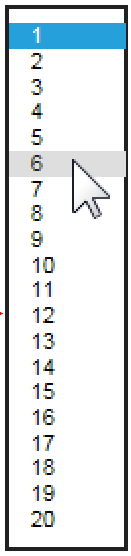

A site can have multiple hunt groups. Each hunt group has a unique phone number, which is provisioned when the services were ordered.



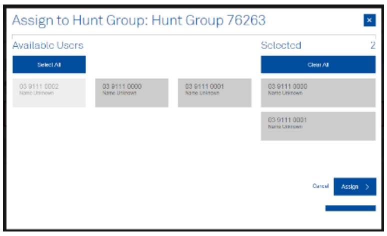

Clicking Edit will open the hunt group pop-up window, similar to that shown below:




The Calling Line ID can be edited as required in this field. Place your cursor in the field and click ‘X’ to clear.



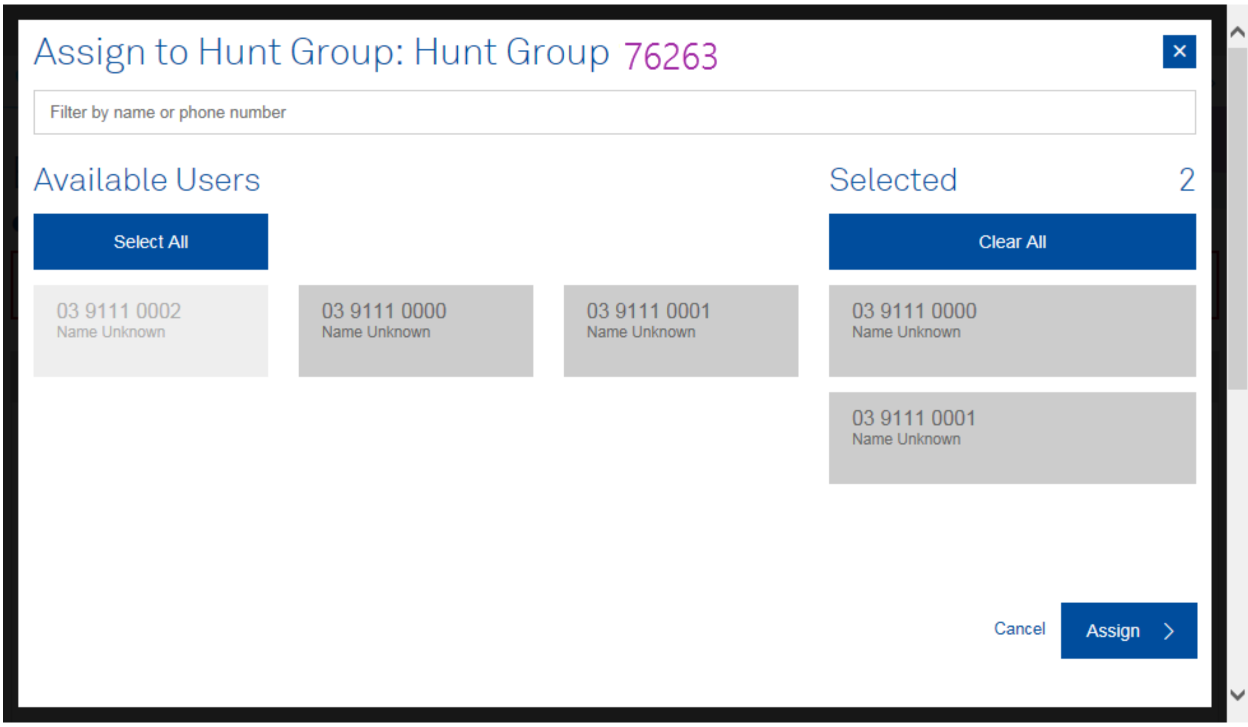

Clicking either of these Change icons will open a pop-up window to allow you to assign/unassigned phone Numbers.




**Note:** Only analogue users/agents can be assigned to a hunt group.

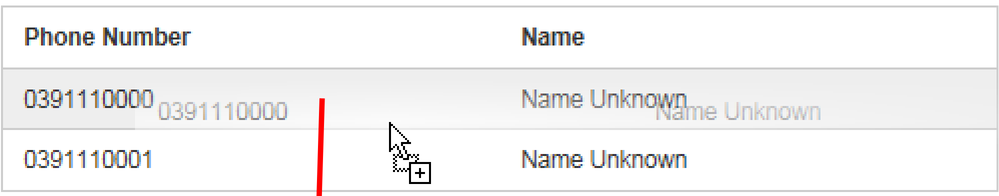
You can also see a list of your analogue users/agents by filtering on user type in the **Users** tab.

To add or remove the agents in a hunt group, click the Edit icon to open the pop-up window:

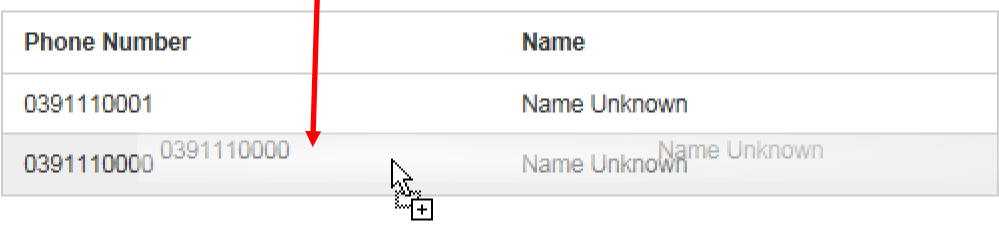



Where the ‘Regular’ option is selected under Call Distribution Policy, the agents will be called in the order presented in the table in the ‘Agents’ section at the bottom of the window. You can reorder them using a drag-and-drop method as demonstrated below, which show 0391110000 being moved to the second position.

1.




2.



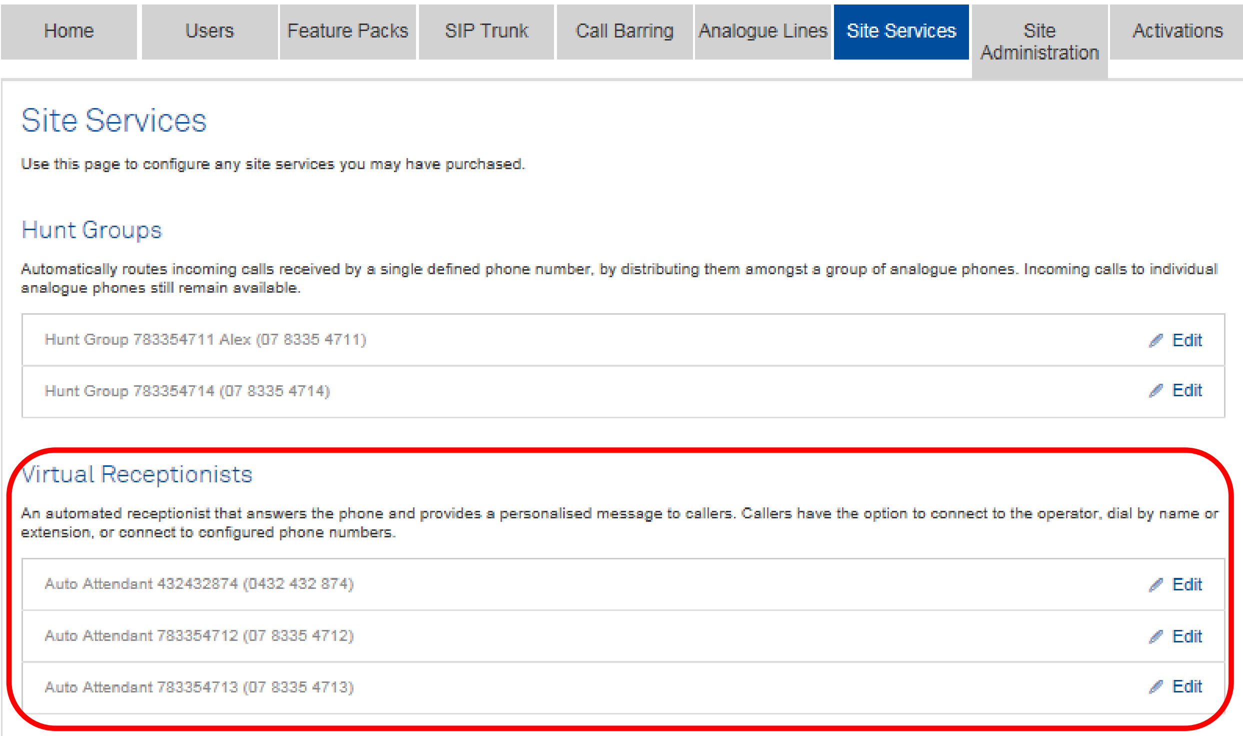
## 3.13. Virtual receptionist (auto attendant) services

The virtual receptionist (auto attendant) is as an optional feature.

It automatically answers incoming calls with a single-level IVR menu structure. Features include:

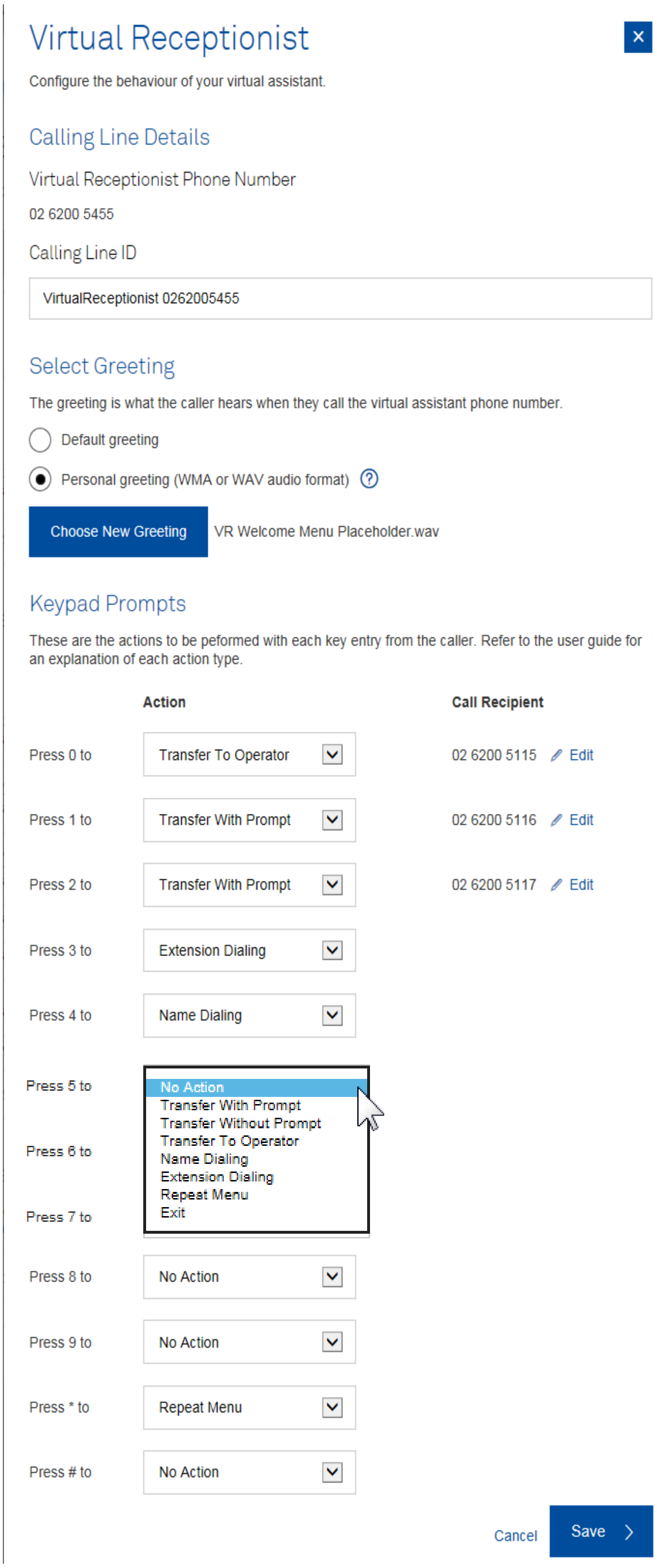
- Personalised greeting to callers
- Option to connect to a receptionist
- Dial by name with phone keypad letters
- Dial by extension number
- Connect to configured destination based on selection from a menu
- Repeat menu option

To access the virtual receptionist service go to the **Site Services** tab.




Clicking the Edit icon opens the configuration pop-up.

The Virtual Receptionist **Calling Line** is the number which needs to be advertised for customers to call to be presented with the auto-attendant greeting and options. This number must be one of your available service numbers.




The greeting is the welcome message and menu prompts, referring to the corresponding keypad numbers.

See next page for more details.

The keypad prompts let you control your calls and aid in the design of the welcome prompt. The welcome message needs to present these options to your callers.

See next page for more details.

The keypad prompts dropdown presents a range of action options to associate with the corresponding keypad digit.

The Virtual Receptionist can dial external numbers and as such will need to be provisioned with its own service number and calling plan.

### 3.13.1. Select greeting

Your Virtual Receptionist needs a greeting that will announce your business and the available options that you have set to manage these incoming calls. You can use a default greeting or record a .wma or .wav file and upload it. See below for specific format requirements.

*For example:*

> *"Welcome to Space Elevators Tidbinbilla*
> *For faults and issues, press 1,*
> *For the accounts team, press 2,*
> *To enter an extension directly, press 3,*
> *To use our automated name directory press 4,*
> *To speak with the operator, press 0,*
> *To listen to these options again, press star."*

**For .WAV files:**
8.000 or 16.000 kHz
8 or 16 bit mono
µ-law, A-law, or PCM

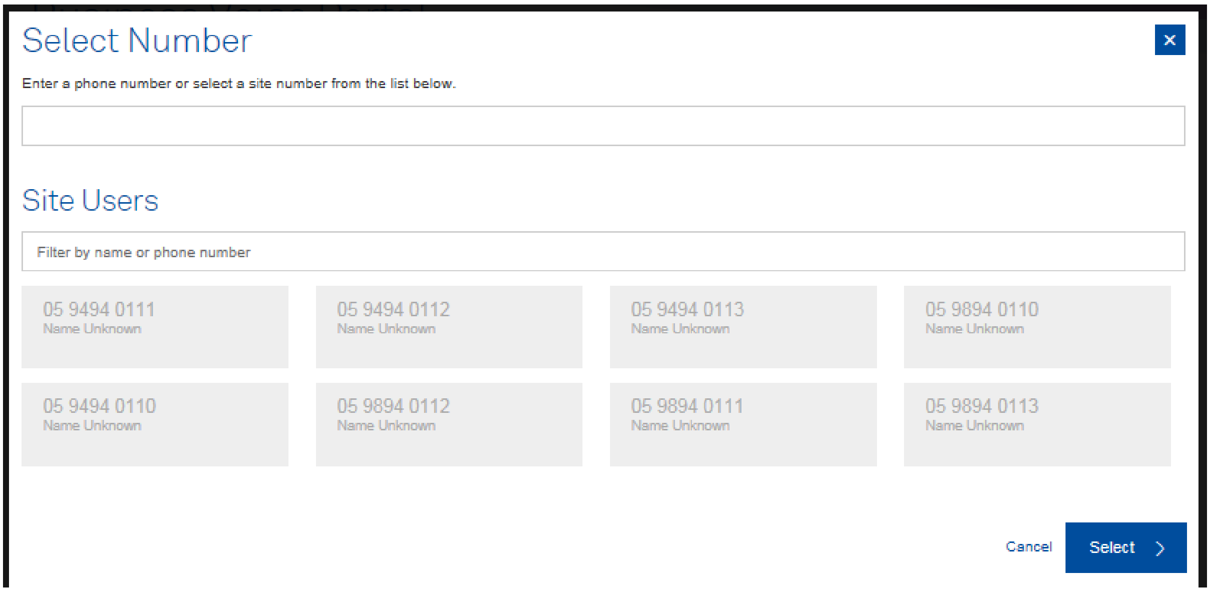
**For .WMA files:**
8.000 or 16.000 kHz
16 bit mono
µ-law, A-law, or PCM

### 3.13.2. Key prompts

Below is the full range of options that can be performed by the caller using their phone keypad.

| Option | Description |
|---|---|
| No Action | Used when the keypad number has no required function (this is the default setting.) |
| Transfer With Prompt | Transfers the caller with a voice message that simply advises them that they're being transferred. Click Edit to enter the number. |
| Transfer Without Prompt | Transfers the caller with no voice message to the number specified. Click Edit to enter the number. |
| Transfer To Operator | Transfers to the operator number specified. Click Edit to enter the number. |
| Name Dialing | Allows a caller to enter a user's name via the keypad numbers e.g. "2ABC", for either A, B or C, and "3DEF" for D, E or F, and so on. |
| Extension Dialing | Allows a caller to dial the extension number directly, if they know it. Useful where there is no indial feature available. |
| Repeat Menu | Replays the menu options. |
| Exit | Ends the call with no action. |

For the options that require a transfer number, clicking Edit opens the pop-up window to select/assign the transfer number.

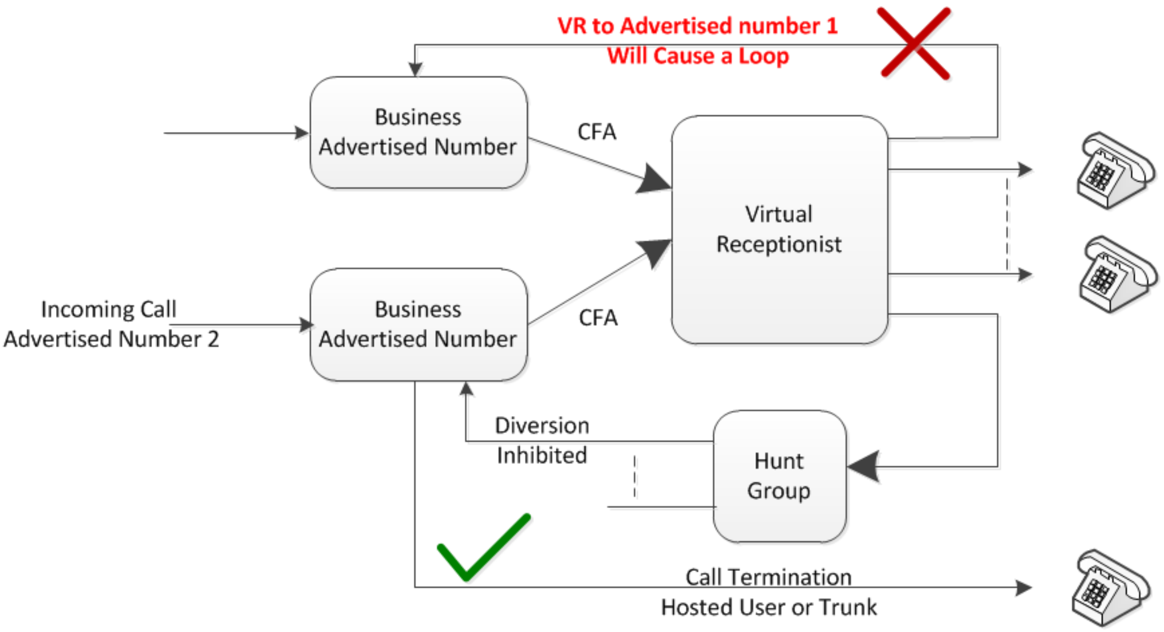



### 3.13.3. Sample implementation of virtual receptionist

Exercise care when developing the flow for the Virtual Receptionist as it is possible for calls to be circular routed (meaning that it can be made to direct calls back on itself, in which case calls would never be answered).

Call Forward Always should be used to route calls from the site's advertised number (and/or service number) to the virtual receptionist.

**Note:** To divert calls to the Virtual Receptionist, the optional Business Line Feature Pack is required as this includes the call forward features required.

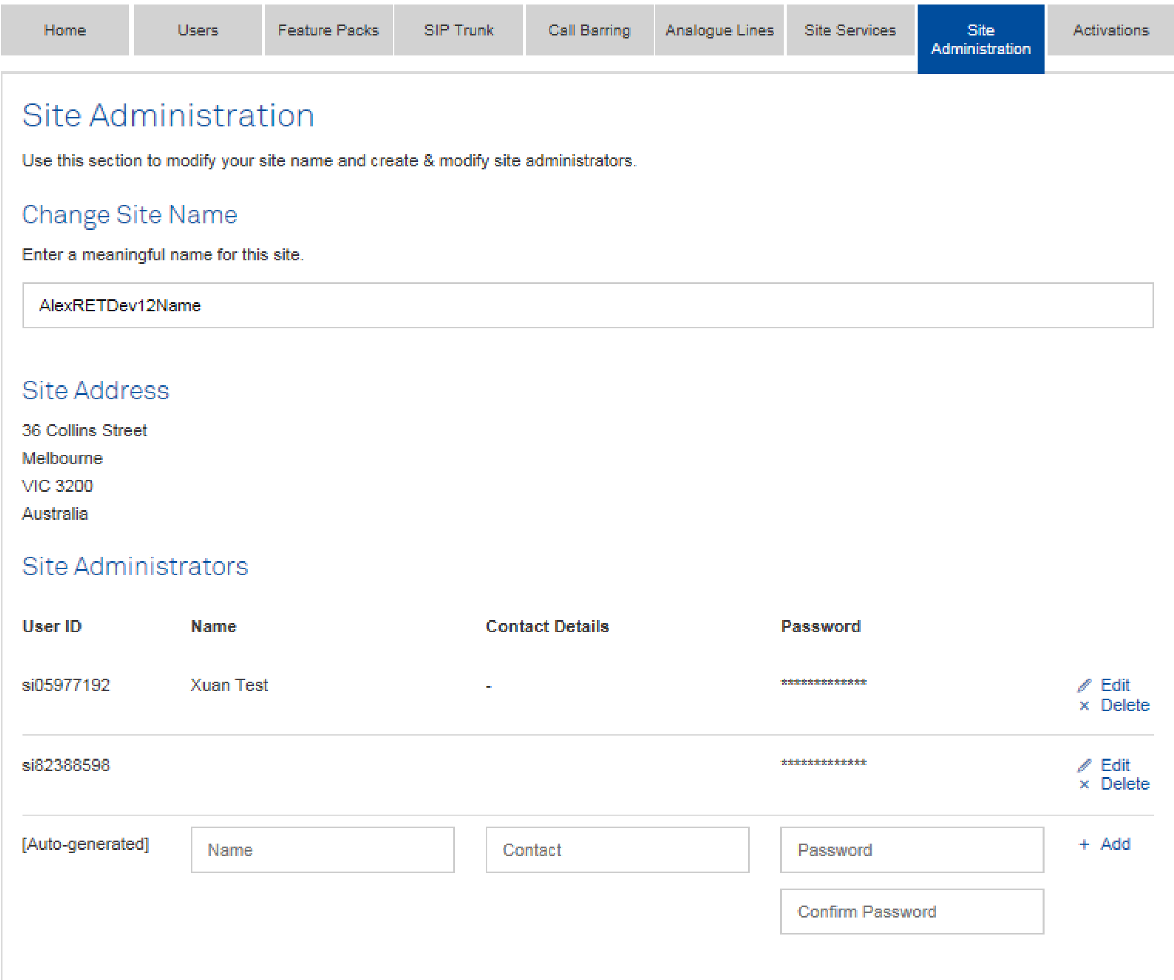



Virtual Receptionist calls terminated within the enterprise are free, but calls terminated outside the site are charged. A calling plan must be linked to the virtual receptionist service, so calls made externally can be billed to the customer.

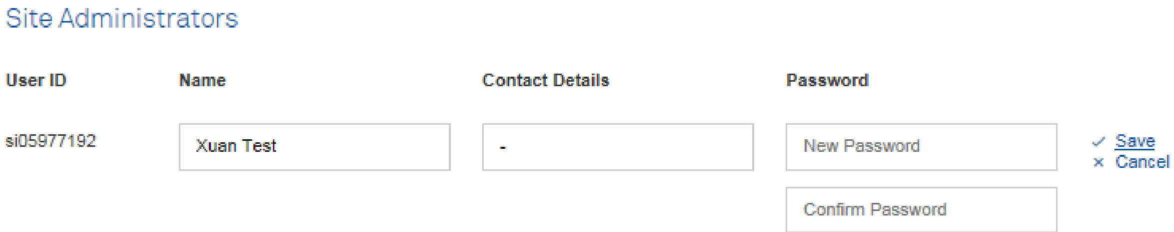
## 3.14. Site administration

The **Site Administration** tab provides access to edit the site name and edit/add/remove site administrators.

**Note:** In this Customer Management Portal, you can create/remove other site administrators.




Click × Delete to remove an existing administrator or the Edit icon which will present these editable fields:




When you've finished making changes, click the ✓ Save icon or × Cancel.

To create a new Administrator click + Add.

## 3.15. Migrations

Migrations relates to the number migration of your services (if applicable) when your product is first installed. The process, represented in the diagram below, involves the coordinated use of the Telstra Business SIP Customer Management Portal and performing test calls *from* and then *to* your business number(s).

The migration is performed in three phases (described below) with an option to go back (roll back), which returns both outgoing and incoming migrations depending how far you may have progressed.

**Important:** When you begin the migration or roll back your services **you must** also move the PBX cabling between the NT1 and SIP NTU accordingly.

Once you've activated the **Outgoing**, **Incoming** *and* **Finalise** phases, the **Rollback** option on the **Finalise** box remains available for up to seven days, depending on how many rollbacks are performed during the migration. This seven-day period allows time to assess the operational performance of your Telstra Business SIP services; however it is recommended that any required **Rollback** be performed within **six (6) days** to avoid losing the Rollback option.

Between the **Incoming** and **Finalise** steps there is a limit of three rollbacks*, after which the rollback option is not presented.

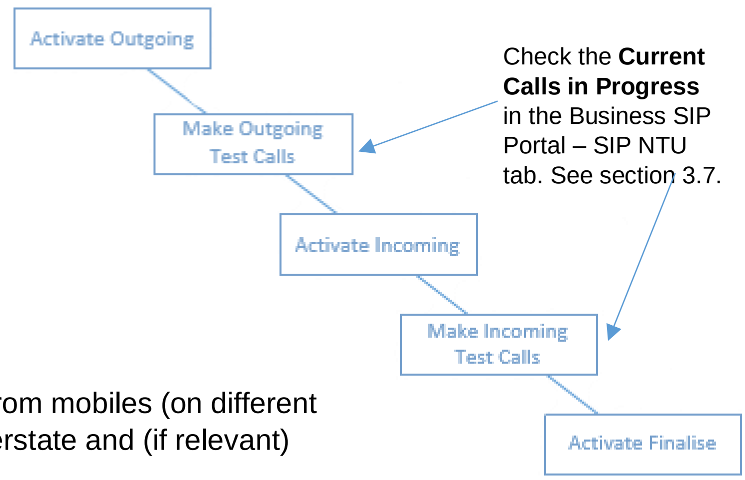
The migration phases are:

1. **Outgoing** – enables the outgoing call capability and allows you to test your outgoing calls, with an option to roll back.
2. **Incoming** – migrates incoming calls and allows you to perform tests, again with an option to roll back.
3. **Finalise** – begins a seven-day trial period to evaluate your services in daily operation, within which time you can either roll back using the Customer Management Portal interface or with assistance from Telstra support, where the rollback option isn't presented. **Note:** Telstra support can only help with rolling back your services during the seven-day 'soak' period.

* The three rollbacks mentioned above are cumulative across phases **2. Incoming** and **3. Finalise**.

During the number migration, test calls are required for you to evaluate whether to roll back or activate the next phase.

**Outgoing test calls** must be made from the handsets connected to the PBX associated with your Telstra Business SIP product. It is recommended that calls be made to different types of networks, including but not limited to, mobile (on different carriers), local standard numbers, interstate and (if relevant) international numbers.



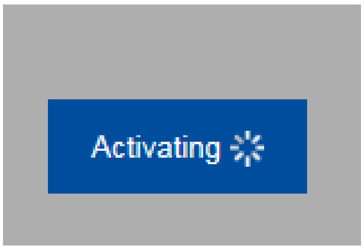

**Incoming test calls** – arrange calls from mobiles (on different carriers), local standard numbers, interstate and (if relevant) international numbers.

Below are the five status indicators you'll see during the migration process.

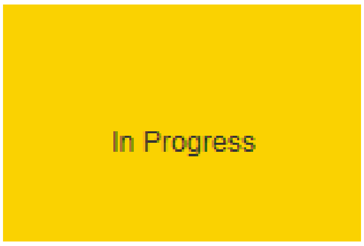
**Note 1:** The second and third states are transitional and may only appear briefly.

**Note 2:** During the process, the web GUI will need to be manually refreshed although the background update will take 5 minutes. It is likely that the initiated changes have taken place and so testing may be started even though the web page indications may not have changed.

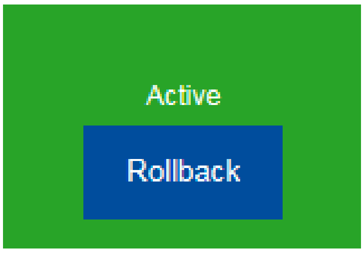

**Pending Phase**
This indicates that the phase (outgoing, incoming or finalise) is ready to start migrating.
Click **Activate** to begin the migration phase.



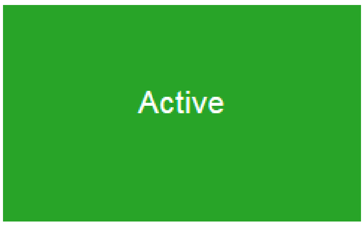

**Phase Initiated**
This status appears momentarily while the migration is started.




**Phase in Progress**
This status appears during the setting up of services in the background.




**Phase Completed – with option to undo**
When this status is presented, you can start testing your calls for the given phase of the migration.
If successful, click **Activate** on the next phase or click **Rollback** to begin again.




**Phase Completed – with <u>no</u> option to undo**
Note: you may have an opportunity to roll back/undo at the finalise phase.

**Note:** Once your seven-day soak period is over or if you perform a total of three rollbacks during the incoming and finalise phases, you will <u>no longer</u> see a rollback option.

These will be repeated for each of the three phases, noting that the second and third are transitional.

### 3.15.1. Starting the migration process

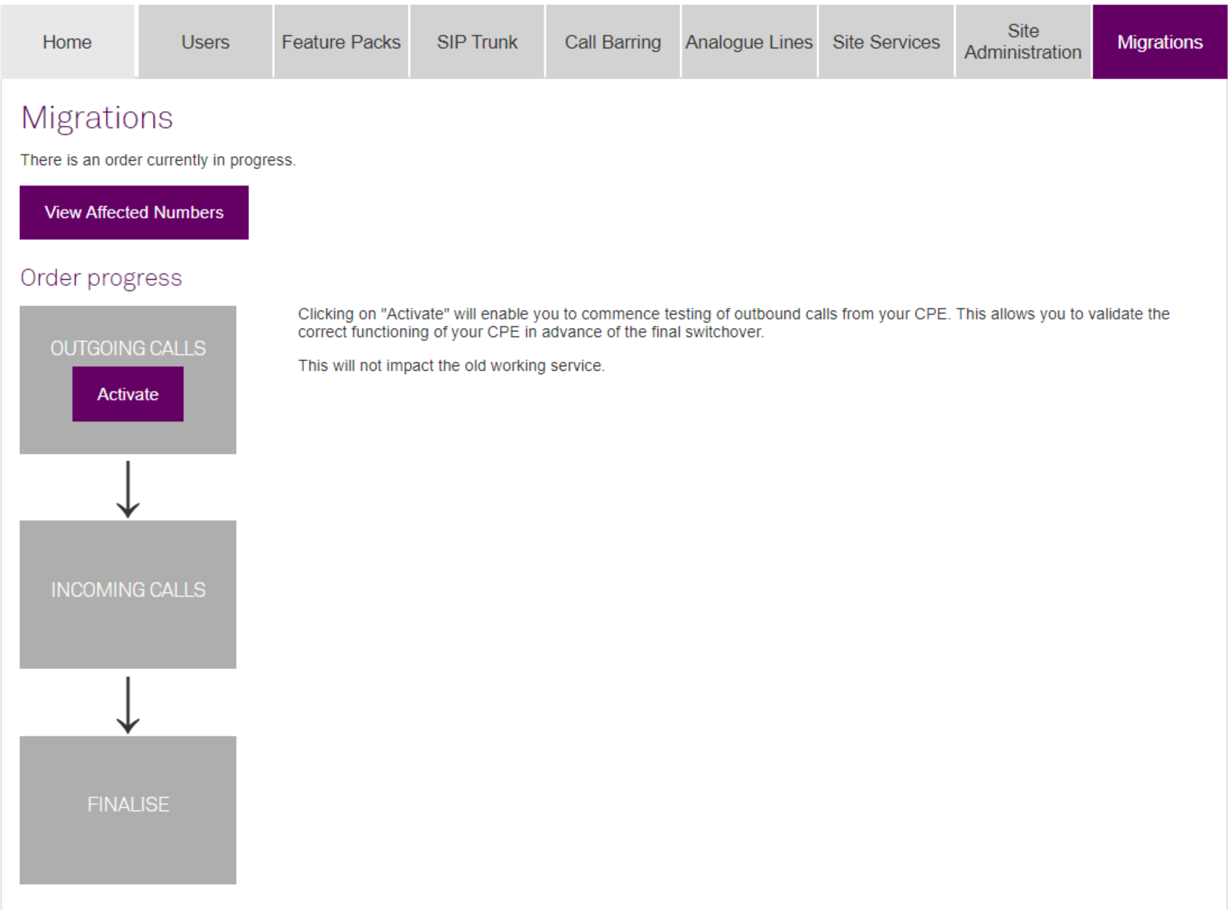
Select the Migrations tab to start the number migration process.

If there are no orders pending you will see the message:

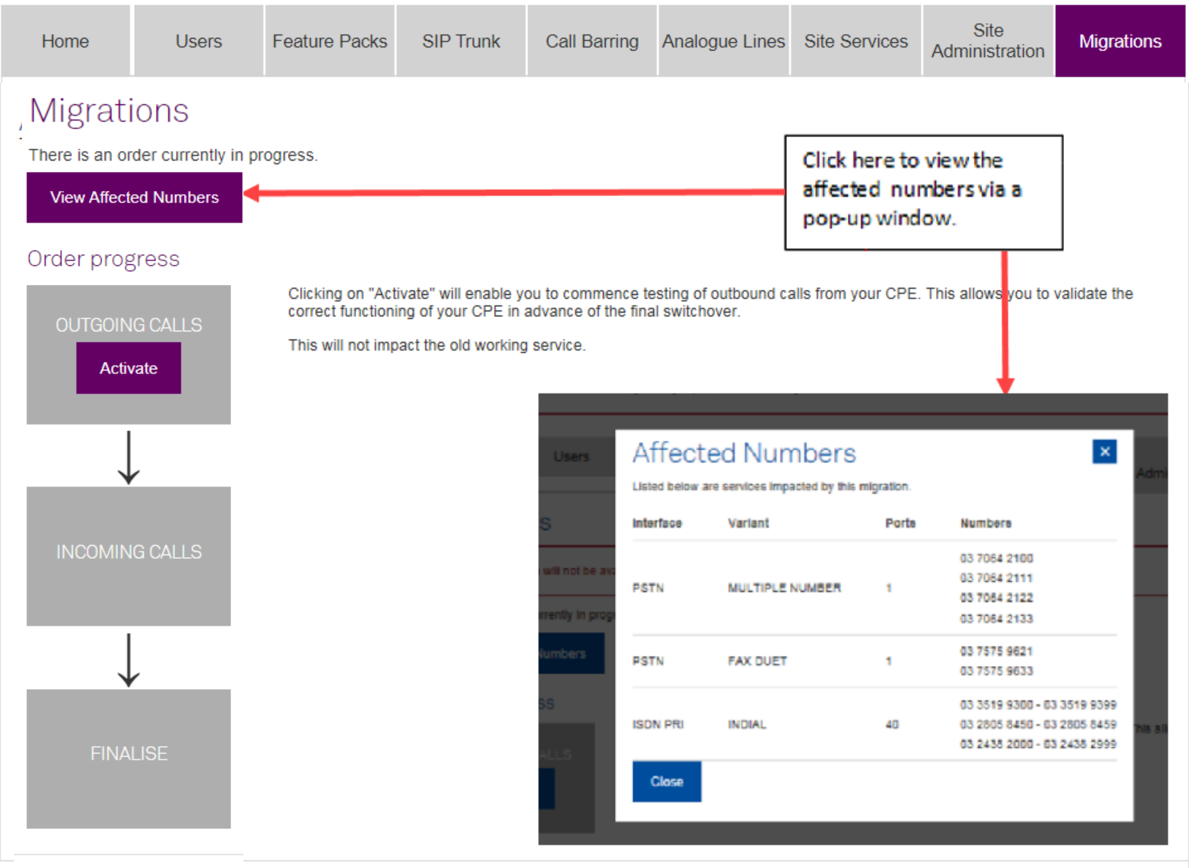
"*You currently have no outstanding orders for number migration*."

Where an order ***is*** pending, the following screens describe:

- how you can check the affected numbers to be migrated
- the process, with text describing the required interactions.

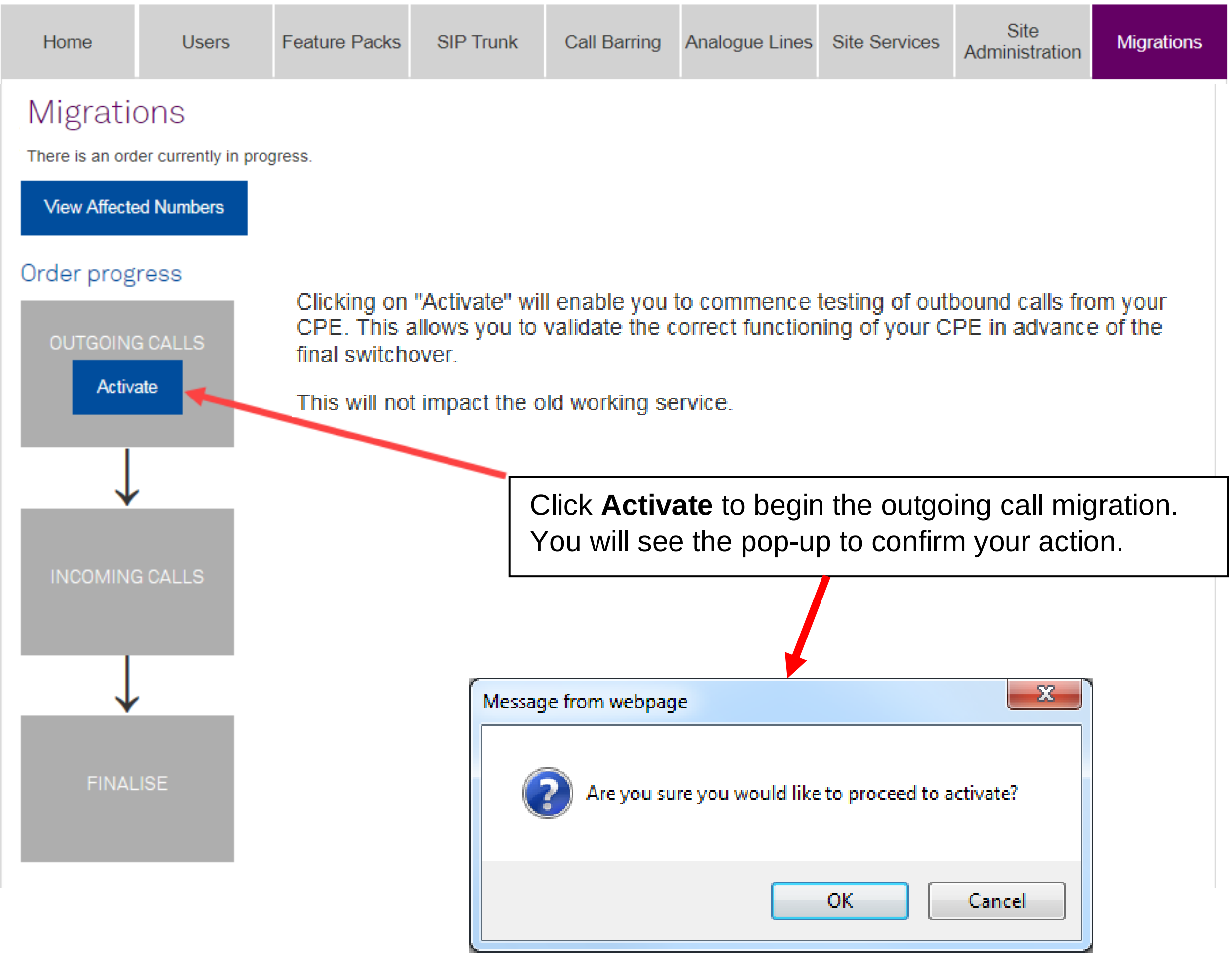


Before you begin the migration, you can view the numbers being migrated by selecting the View Affected Numbers button as shown below. If these numbers are not what is matching your order, contact Telstra or your nominated Telstra Partner.

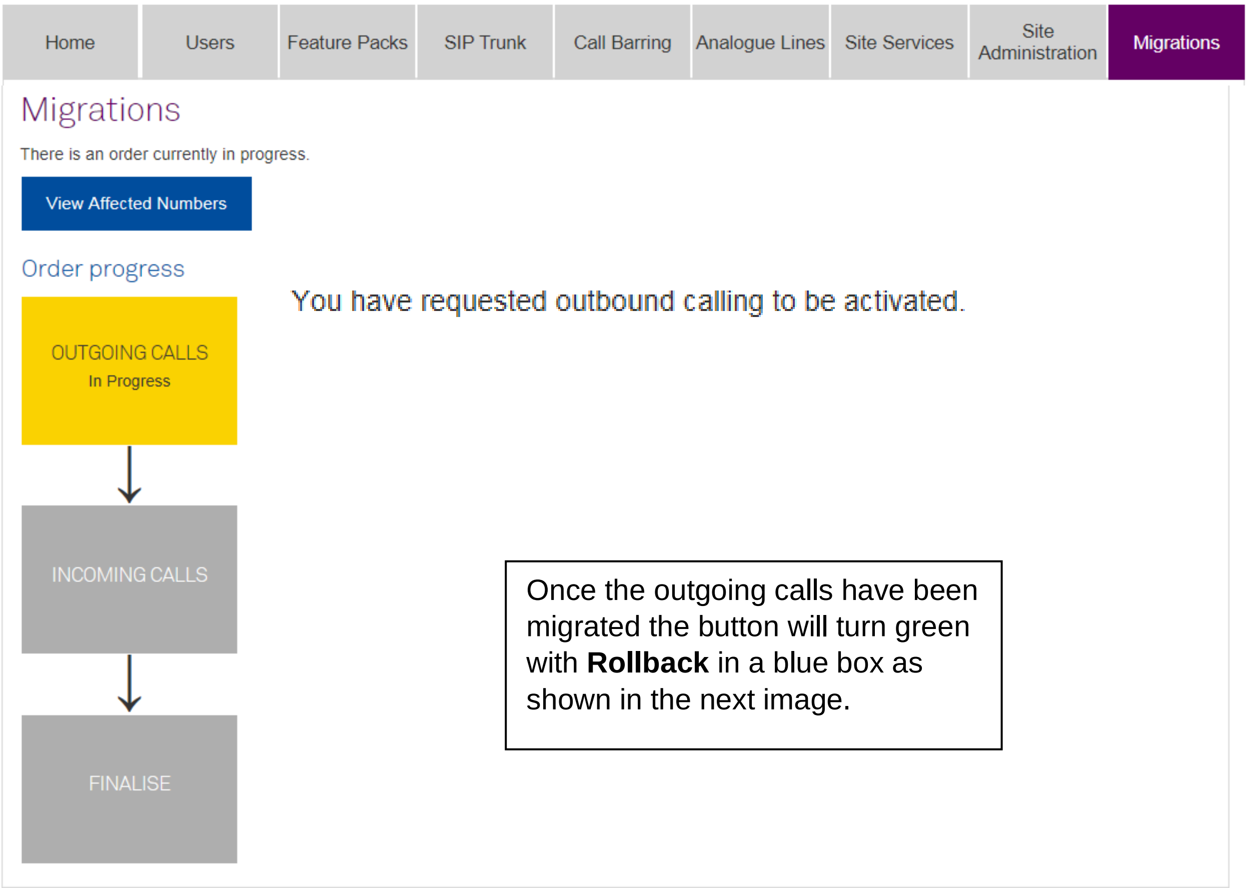



Once you've reviewed this information click Close

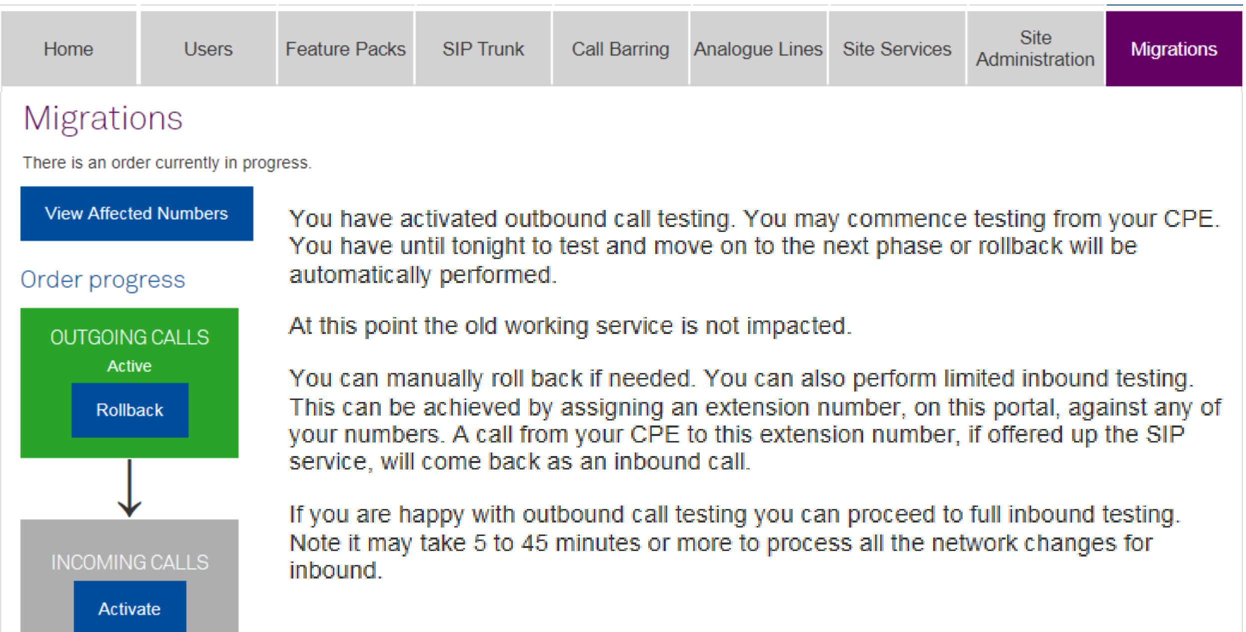
Before you start your migration you'll see this screen with **Activate** in the **Outgoing Calls** box as shown below:

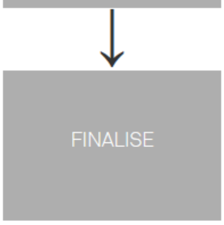


The outgoing calls box will appear yellow for a brief period as the changes are made in the background.



Once the outgoing calls have been activated you'll see the **Rollback** option appear as shown below, plus the **Activate** option on the incoming calls. Using PBX handsets, perform outgoing test calls to landlines, mobiles, interstate numbers (the types you'd use during normal operation).



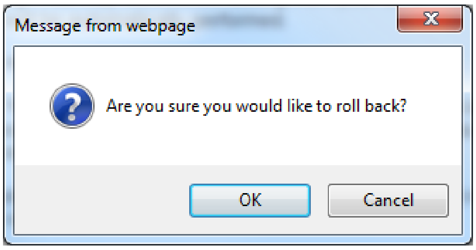



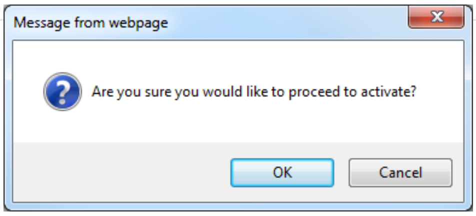

So at this point you have **two** choices:

1. Click **Rollback** to undo the outgoing call migration and you will see this message:

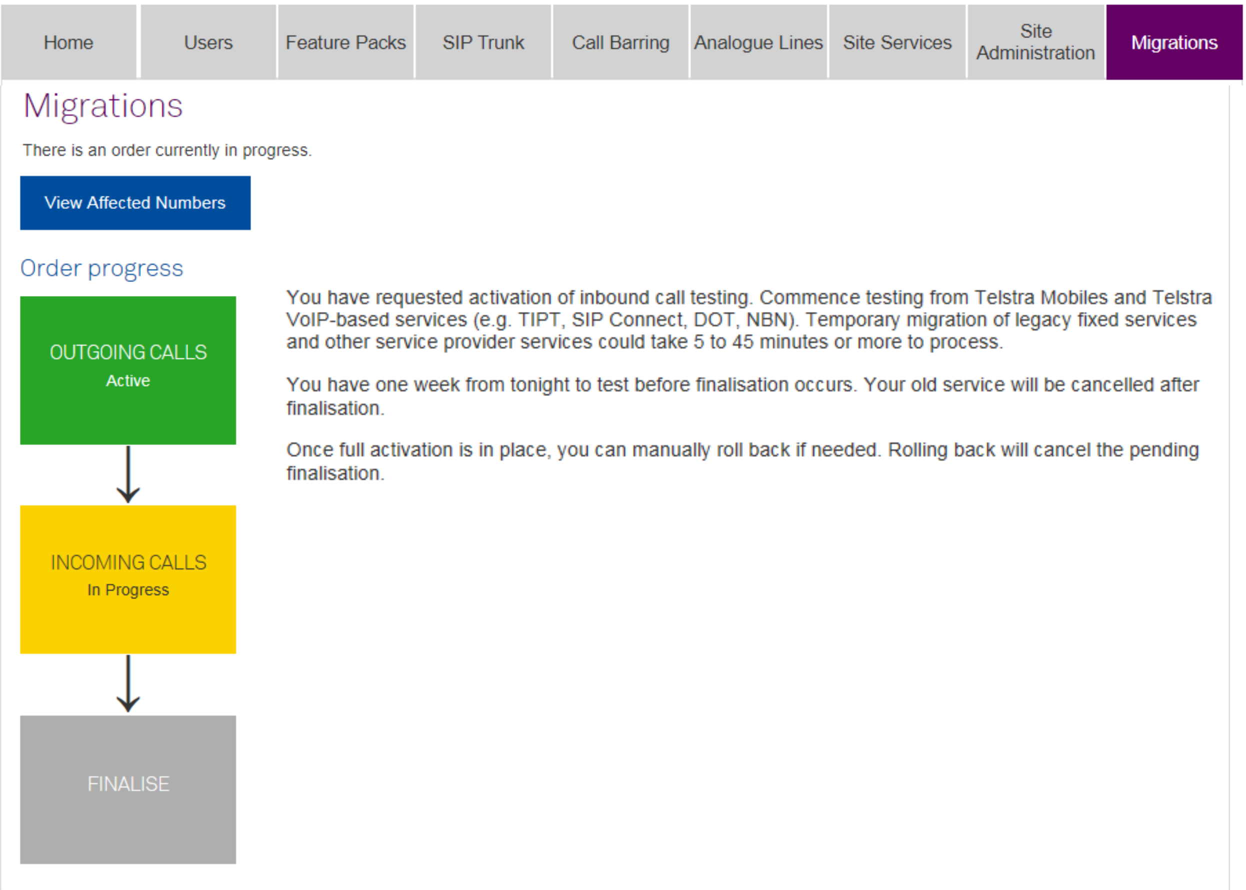
Or

2. Click **Activate** to begin the migration of the incoming calls, in which case you will see this message:



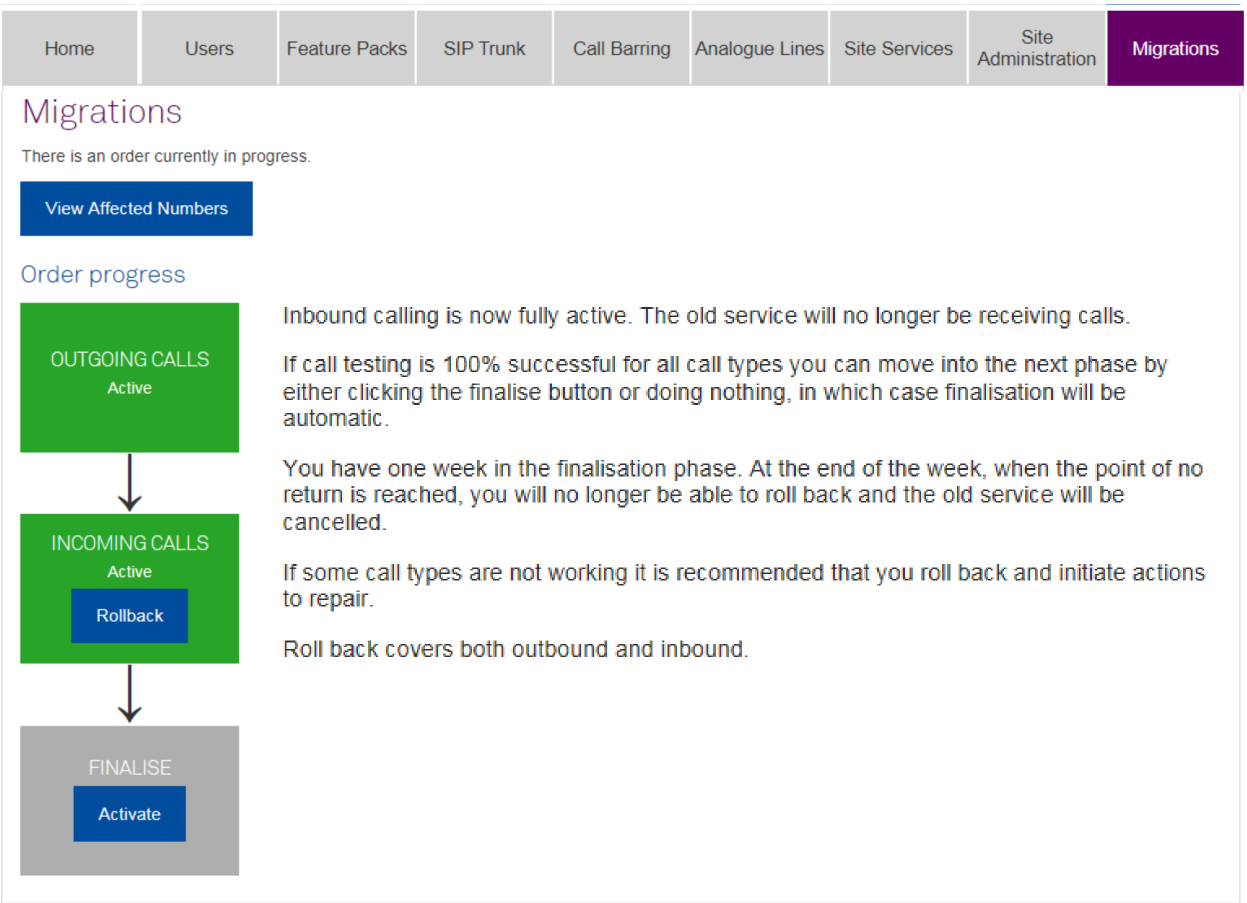



Selecting to **Activate** migration for incoming calls presents the **In Progress** (transitional) status as shown below:

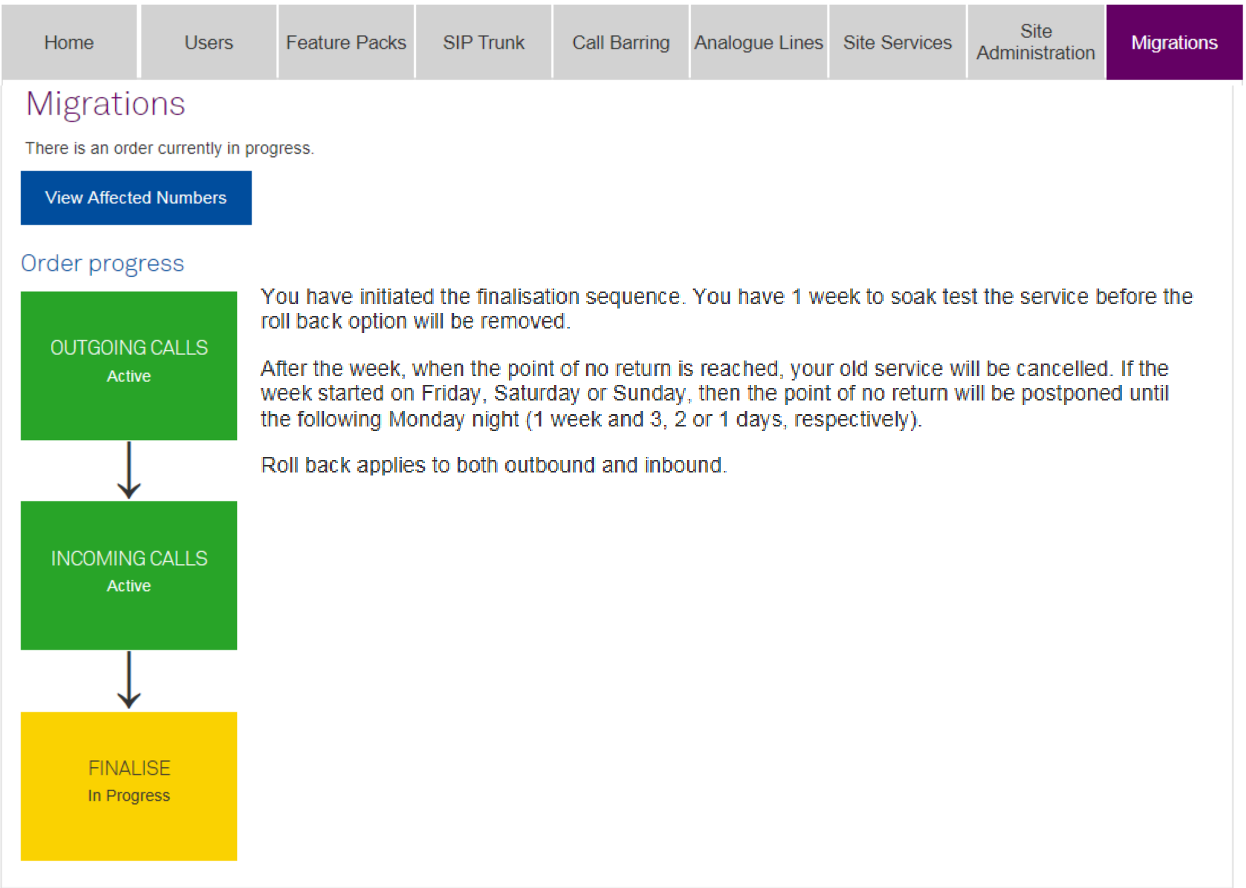



**Note:** The **Activating** (transitional) status has been omitted for clarity.

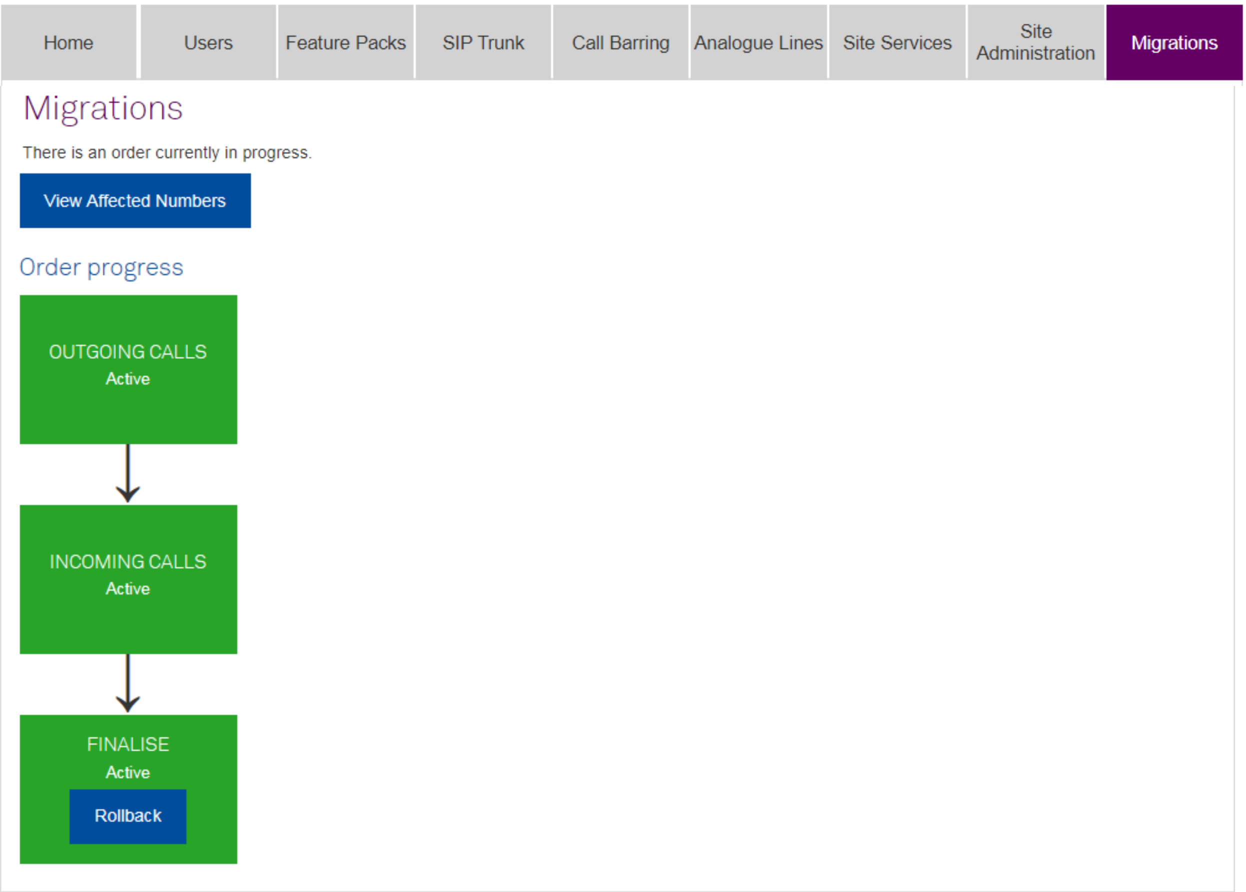
Once the incoming calls have been activated you'll see the **Rollback** option appear as shown below, as well as the **Activate** option on the **Finalise** step. Using external services including mobiles from various carriers, landlines, interstate landline residential and business numbers, arrange calls into the PBX:



Selecting to **Activate** migration for **Finalise** presents the **In Progress** (transitional) status as shown below:




**Note:** The Activating (transitional) status has been omitted for clarity.

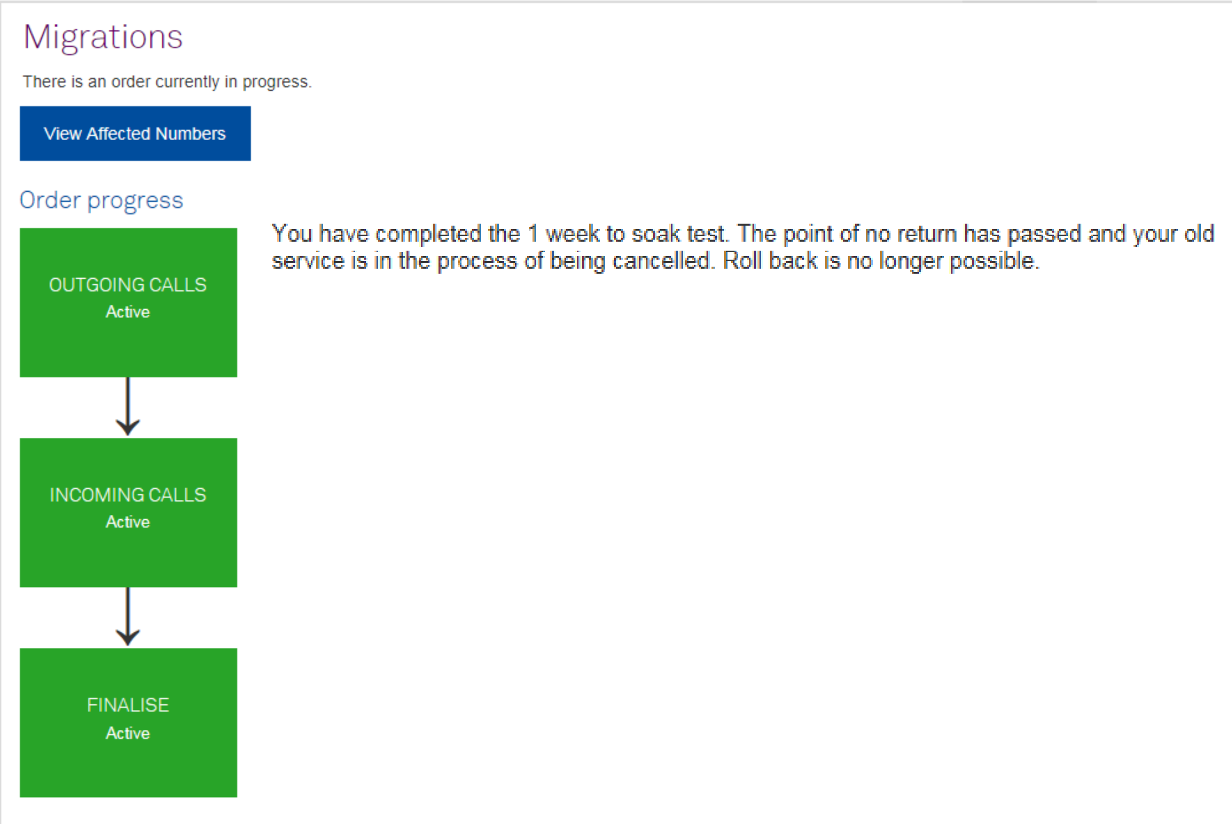



At this point your services are fully migrated; however once **Finalise** goes to the **Active** state, the **Rollback** option is presented for seven days.

**Note:** If you've already used the rollback function three times during the incoming and finalise phases, the rollback option will not be presented.

If you need to roll back your services during the seven-day period, contact Telstra or your nominated Telstra Partner.

At the end of the seven-day soak period your migration will be complete and you will be advised as presented below:




When you see this screen, your old services will be decommissioned.

# 4. User features

Each user has control over their personal details (name and password, covered earlier) via the Customer Management Portal plus:

- Outgoing call ID – calling line identification (CLI) blocking
- Incoming call features including call waiting, call forwarding (where made available) SIM ring
- Voicemail (where made available)
- Remote office.

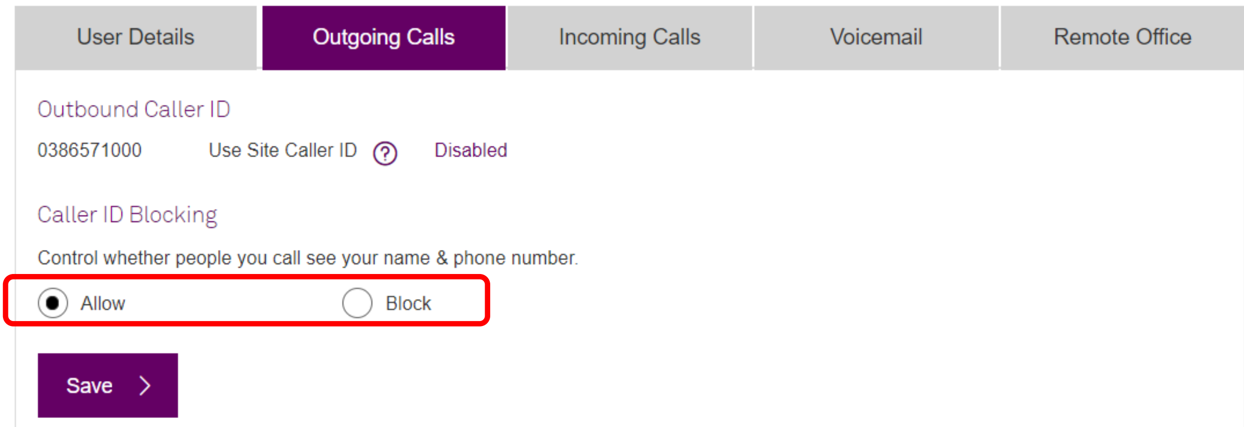
While the features listed *a*bove appear the same for an administrator, the name/password details are quite different, so you may need to refer to the Customer Management Portal User's Guide, when helping users to update their own details.

## 4.1. Calling line identification (CLI) blocking

The CLI blocking feature shows or hides the user's line number to/from the people they call.

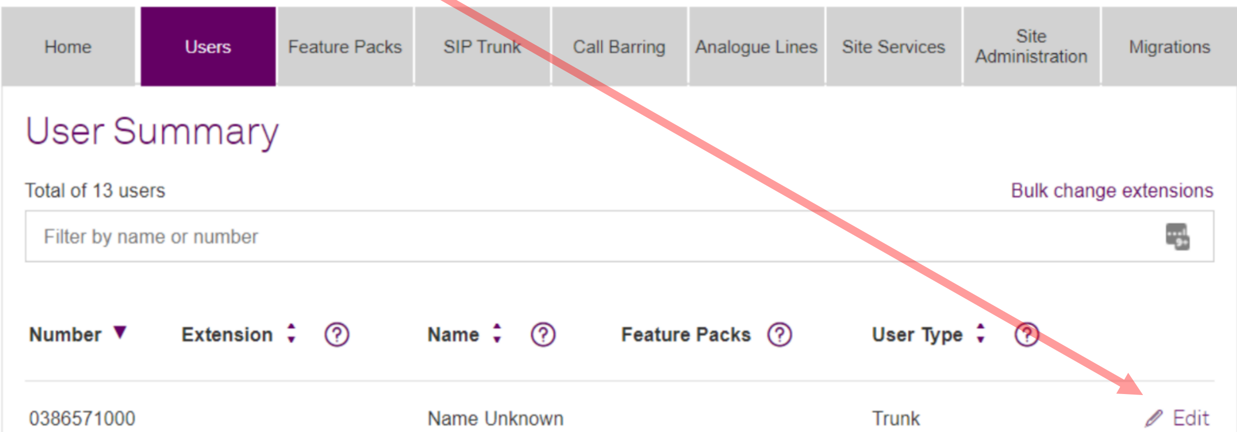
To toggle this feature on or off:

1. Select the **Outgoing Calls** tab.
2. Select the **Allow** or **Block** option as required.
3. Click Save

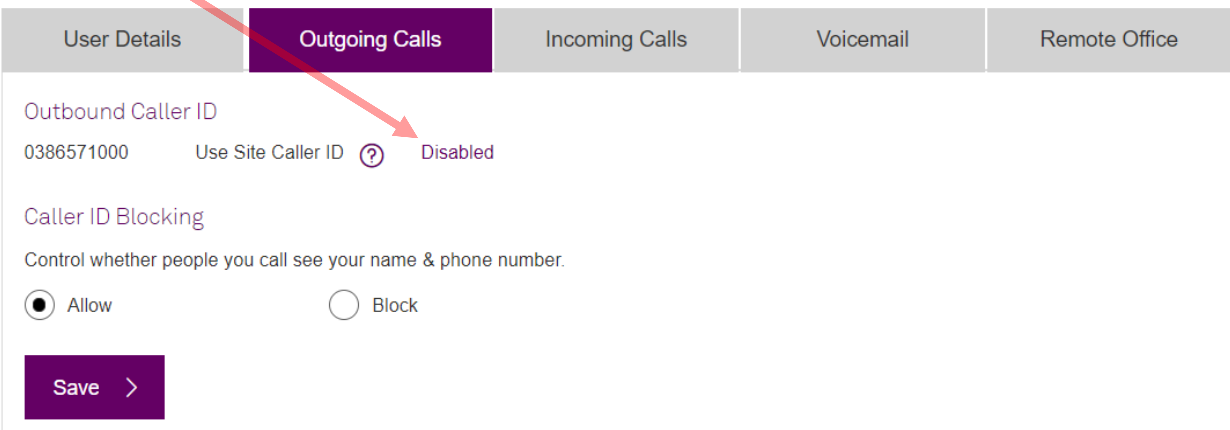


## 4.2. Site Caller ID

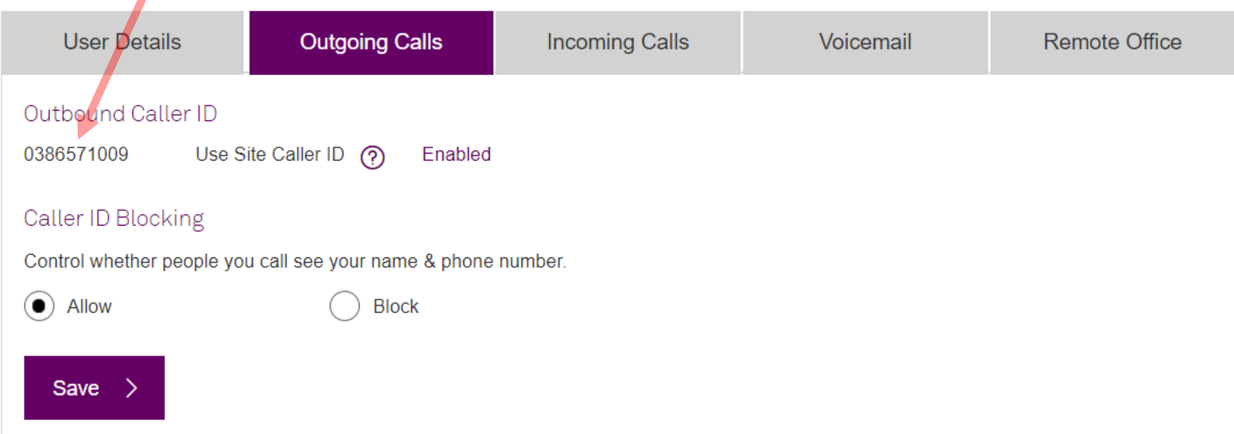
To use the site's designated Site Caller ID as the caller ID for calls made from this number, click on the "Users" tab, then click "Edit" for the number of interest.




Then click on the "Outgoing Calls" tab, then enable the "Use Site Caller ID" feature by clicking on "Disabled" and select "Enabled", then "Save".




The "Outbound Caller ID" field will then display the number that will be used as CLI for outbound calls. Only one number on a site can be assigned as the Site Caller ID. Change the Site Caller ID on the "Home" tab (ref section 3.1).



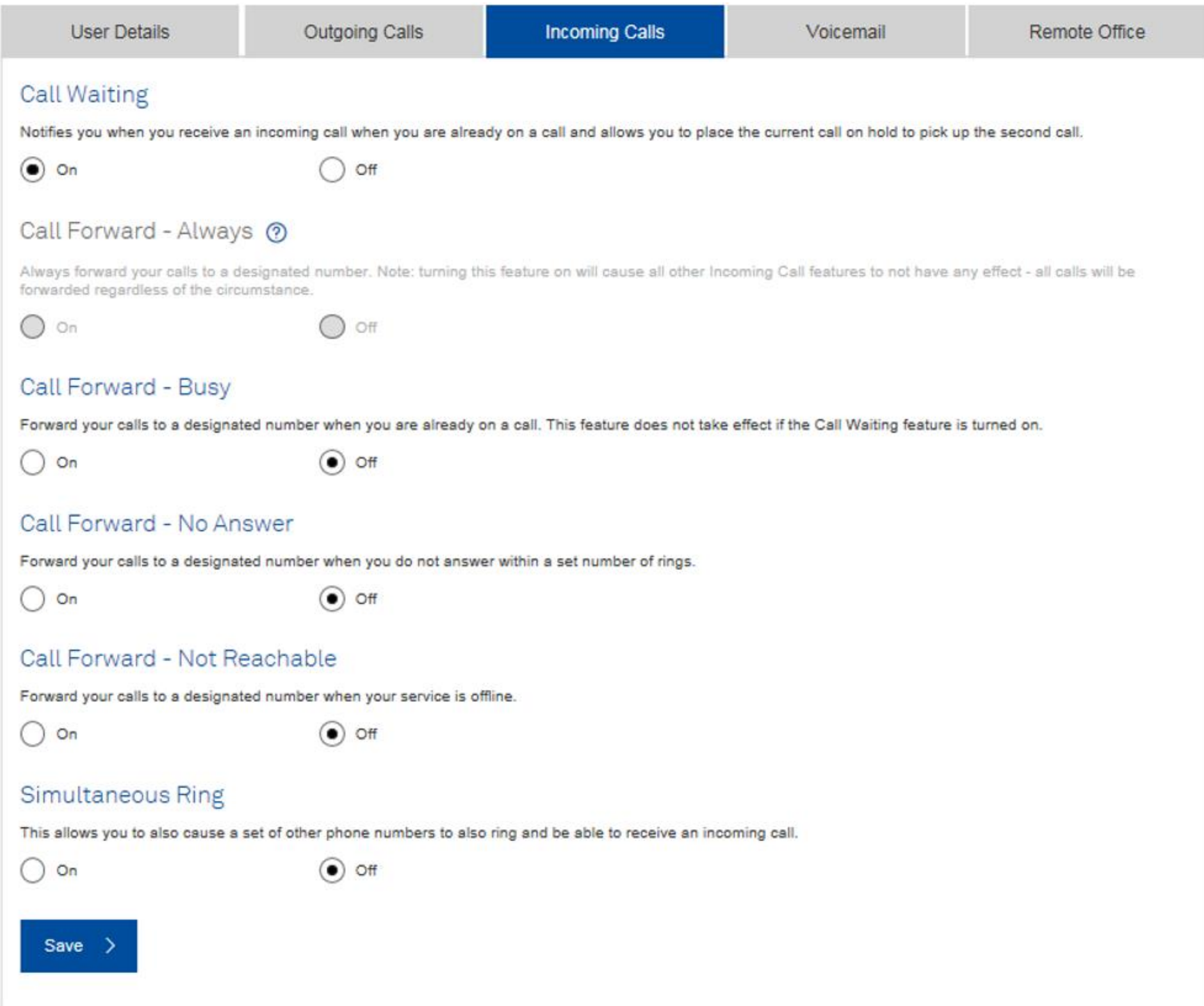
## 4.3. Incoming call features – activate/deactivate/configure

There are a number of features under the **Incoming Calls** tab:

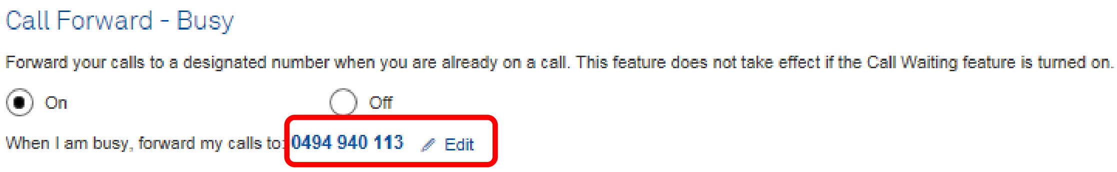
- **Call waiting**
- **Call forward** including – (**always**/**busy**/**no answer**/**not reachable**)
- **Simultaneous ring**.

Depending on the feature packs purchased, any features you don't have access to will be greyed out and accompanied by a ? symbol to indicate that help is available, as demonstrated below next to **Call Forward – Always**.

Where a feature is available, simply select to activate or deactivate it and click **Save**.




Where a feature is activated and there is an Edit icon, you are able to make changes to the settings for that feature. For example:



## Incoming call features in detail

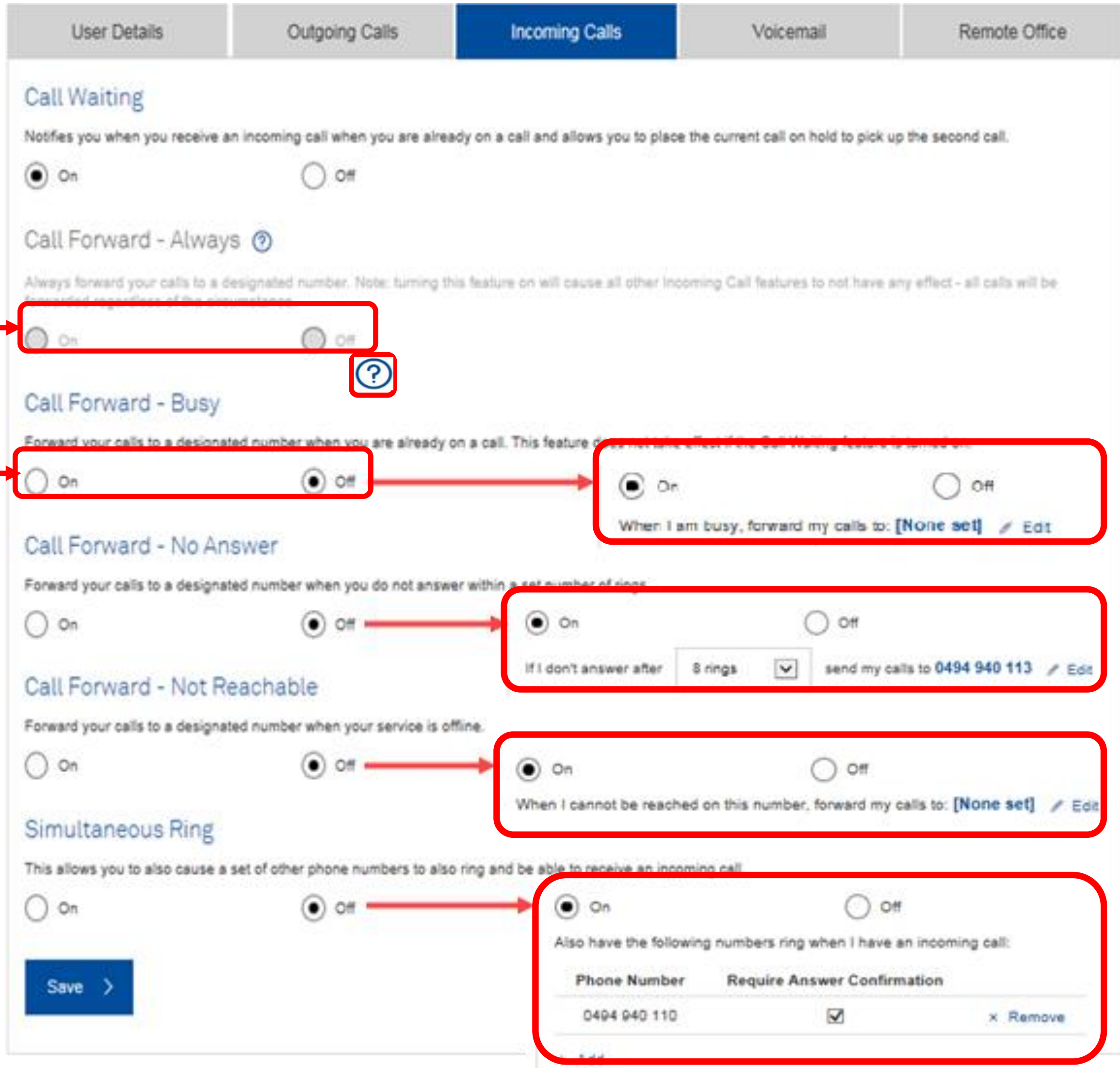
This page presents the features associated with your incoming calls.

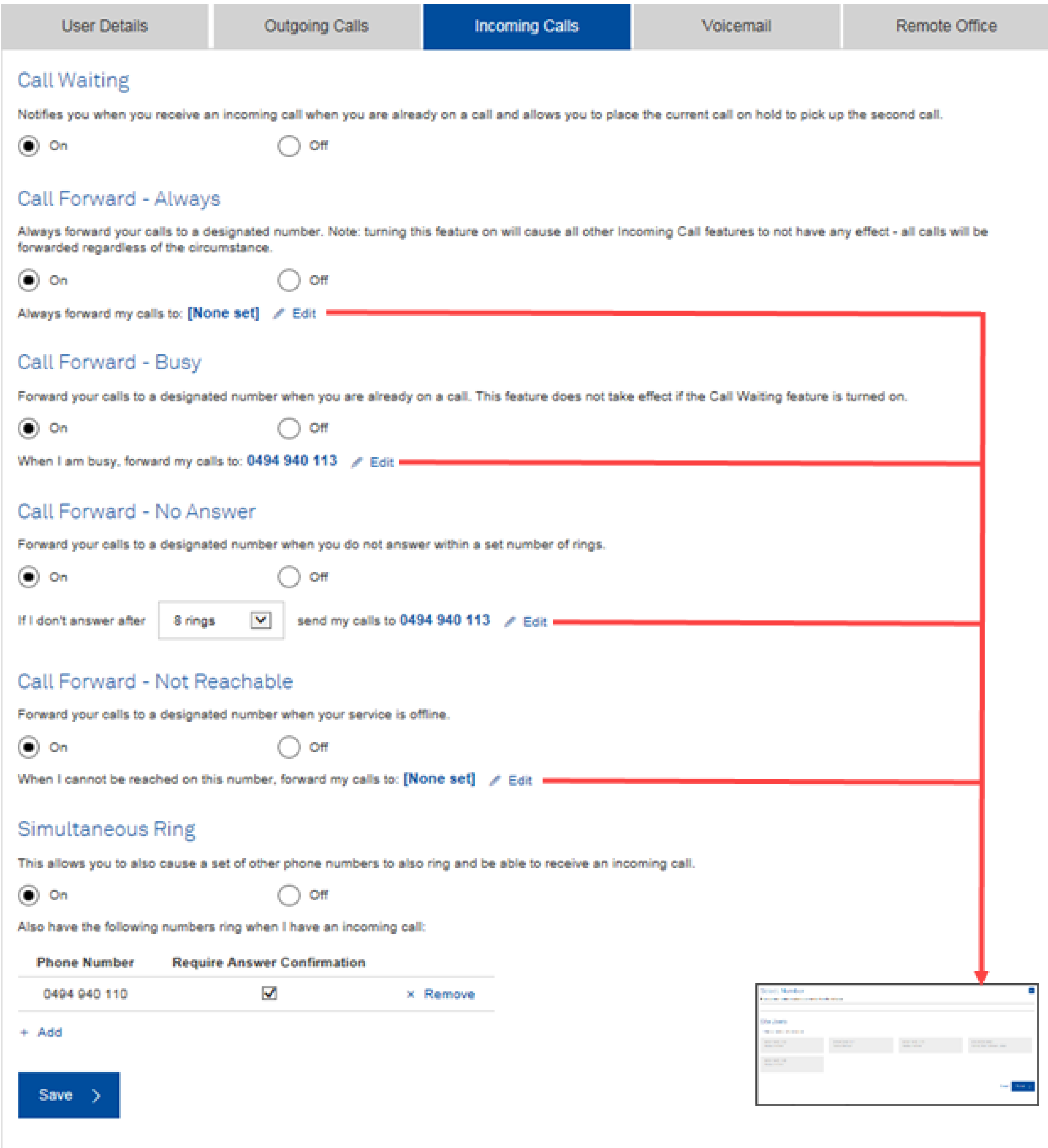
These features can be enabled or disabled using the radio buttons.

Some may be unavailable (greyed out) as the given feature may not be provided with your purchased Feature Pack(s).

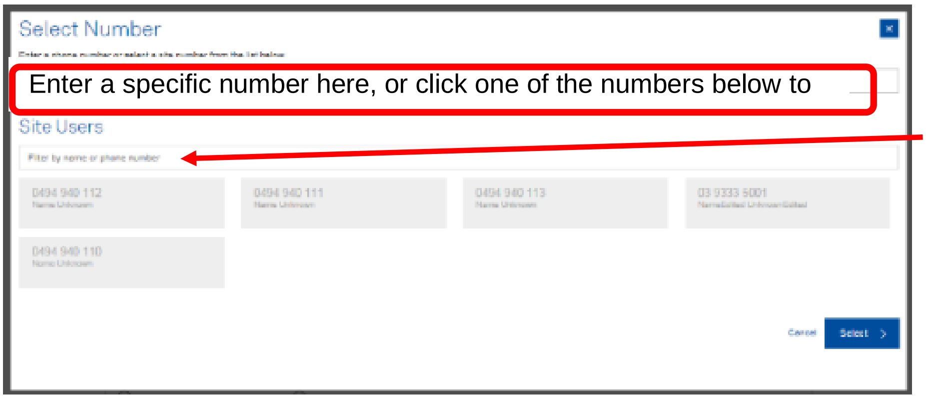
Remember, the ⓘ symbol provides helpful tips and in this case advises:

> You do not have the necessary Feature Pack to you to use this feature. Please contact your administrator to enquire about enabling this feature.




User Details
Outgoing Calls
Incoming Calls
Voicemail
Remote Office
Call Waiting
Notifies you when you receive an incoming call when you are already on a call and allows you to place the current call on hold to pick up the second call.
On
Off
Call Forward - Always
Always forward your calls to a designated number. Note: turning this feature on will cause all other Incoming Call features to not have any effect - all calls will be forwarded regardless of the circumstance.
On
Off
Always forward my calls to: [None set] Edit
Call Forward - Busy
Forward your calls to a designated number when you are already on a call. This feature does not take effect if the Call Waiting feature is turned on.
On
Off
When I am busy, forward my calls to: 0494 940 113 Edit
Call Forward - No Answer
Forward your calls to a designated number when you do not answer within a set number of rings.
On
Off
If I don't answer after 8 rings send my calls to 0494 940 113 Edit
Call Forward - Not Reachable
Forward your calls to a designated number when your service is offline.
On
Off
When I cannot be reached on this number, forward my calls to: [None set] Edit
Simultaneous Ring
This allows you to also cause a set of other phone numbers to also ring and be able to receive an incoming call.
On
Off
Also have the following numbers ring when I have an incoming call:
Phone Number
Require Answer Confirmation
0494 940 110
× Remove
+ Add
Save

Each of the edit buttons above will present a pop-up where you can enter a specific number or click on any of the numbers displayed to use the details for that site user.




If there are a large number of site users in this list you can use the filter field to show only those numbers beginning with the digits you need.

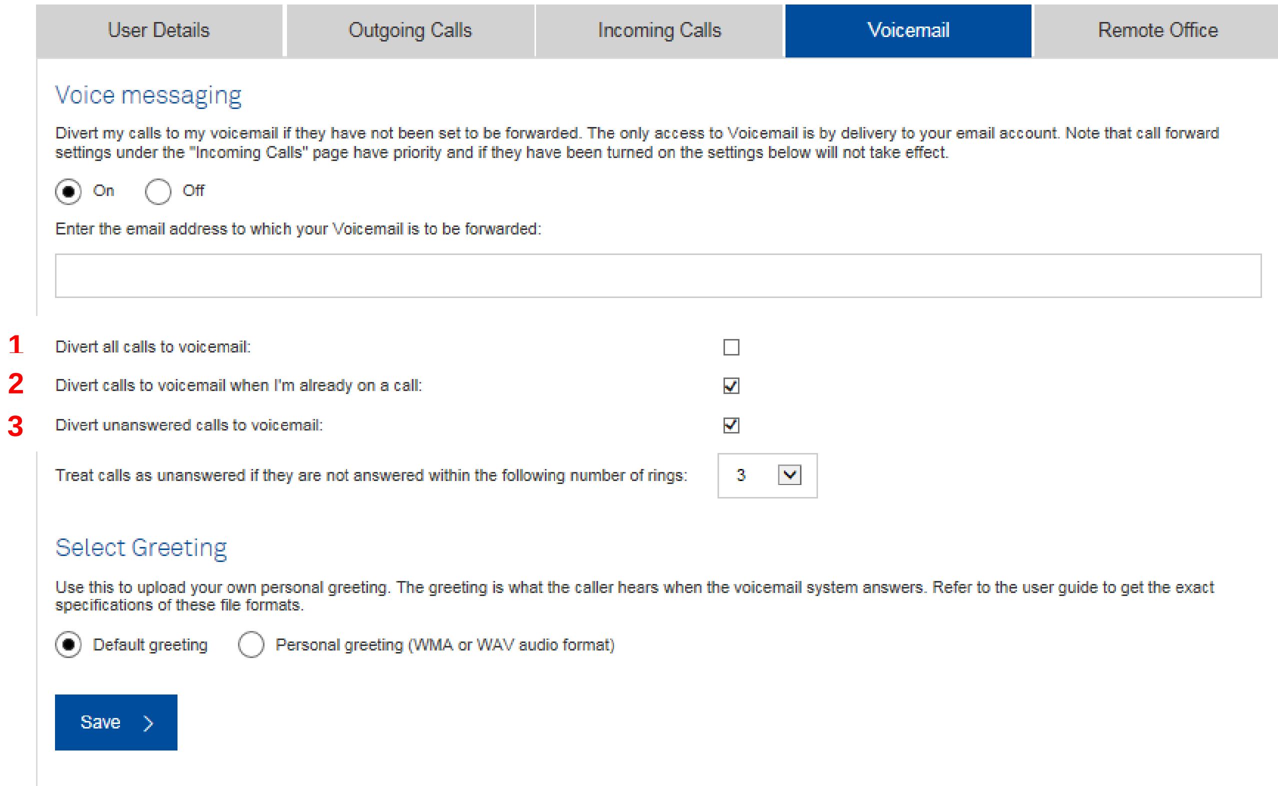
## 4.4. Voicemail – activate/configure

The Voicemail feature allows calls to be diverted to an email address of your choice.

To activate this, select the radio button, which opens the fields to enter the email address and choose which calls you want diverted and under what conditions.

Your three options are:

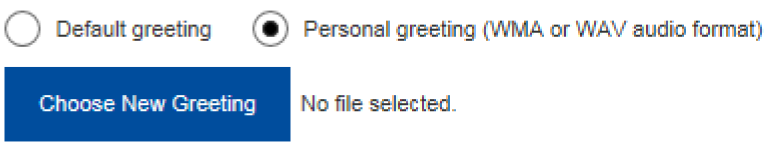
1. Divert all my calls.
2. Divert my calls when I'm (busy) on a call.
3. Divert my calls when I don't answer after your required number of rings.




You can also select to use a **Default greeting** or **Personal greeting** using the radio buttons as shown.


Where you select **Personal greeting** you can upload a WMV or WAV file.

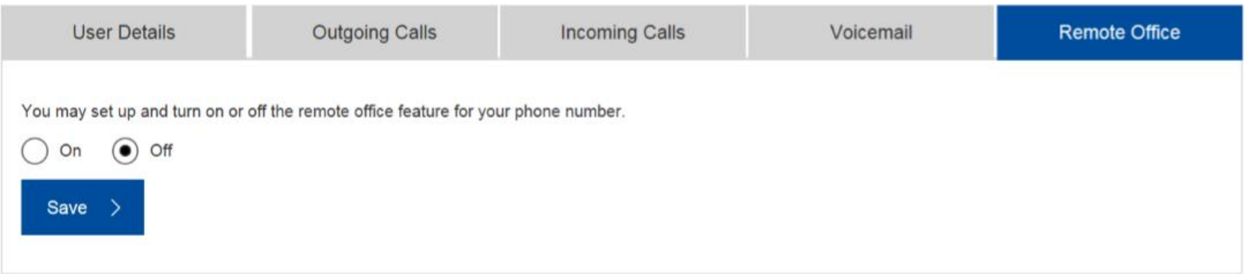
Click the Choose New Greeting button and locate your file.




Remember to click Save.

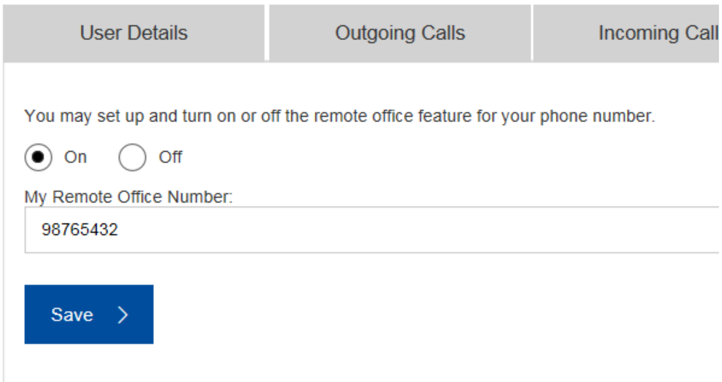
## 4.5. Remote office – activate/deactivate

The remote office feature allows a user to use their home phone, mobile phone or even a hotel phone as their business phone. They can make phone calls from this remote phone and have them billed to your business. The calling line ID that a caller sees is their primary (desk) phone number. This service also directs all calls coming to their business phone to ring the remote office phone.




To activate/deactivate your remote office:

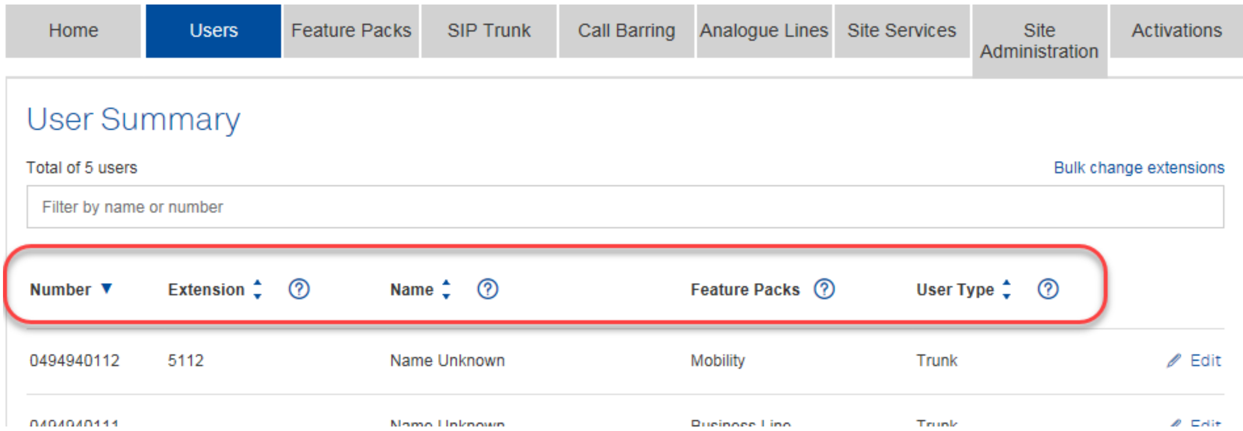
1. Go to the **Remote Office** tab.
2. Select the required radio button.
3. Enter your number.
4. Click Save.



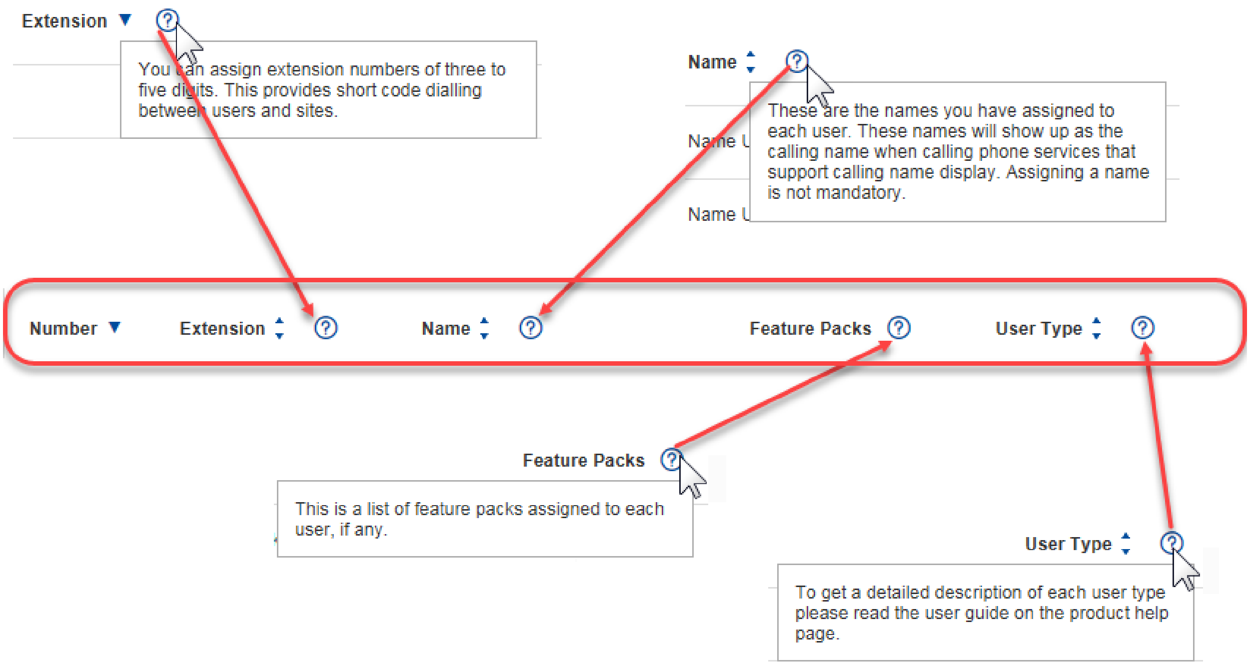
# 5. Help

The Customer Management Portal has a context help system using the ⓘ symbol to indicate that information is available for the item immediately adjacent.

For example, on the users tab you will see a ⓘ symbol next to most of the column headings.




When you hover your mouse over the ⓘ, help text appears for each as shown below:




## 5.1. Notification messages

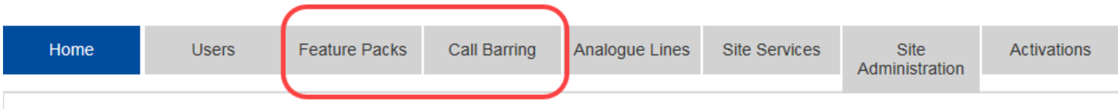
A notifications pop-up may appear to advise of system delays and planned maintenance that impacts the Customer Management Portal.

## 5.2. Dynamic page content

A standard feature of web-based services is the ability to dynamically change to present information depending on the applicable information, features or services.

The Business SIP Customer Management Portal may present pages with different tab content as can be seen below in the example, which doesn't show the SIP Trunk tab, which would be applicable for a service that doesn't have SIP trunks.

Here the SIP Trunk tab is not shown because the associated service hasn't been ordered with this option in this example:




## 5.3. Help link

Click on the "Help" link at the top of any page to get access to the product documentation and for the support phone number.

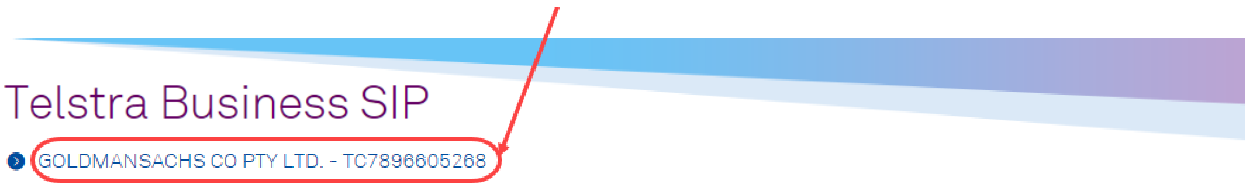

## 5.4. For more help

There are details on your Telstra Business SIP Customer Management Portal that may be requested by the support technician.

The information below in **bold** will help you locate these details:

**The Full National Number (FNN) for your business** – these numbers appear in the user tab.

**Business name** as it appears in the breadcrumbs on your Customer Management Portal **Home** page.




**Site name** as it appears in the breadcrumbs on your Customer Management Portal **Home** page.


**SIP NTU device details** – the make and model of the registered SIP NTU device can be found in the SIP trunk/ SIP NTU tab on the Telstra Business SIP Customer Management Portal.

**Analogue device details** – where applicable the make and model of any device(s) that support(s) analogue services can be found in the analogue lines tab on the Telstra Business SIP Customer Management Portal.

The associated device credentials can be found on each of the respective Customer Management Portal tabs above.

## 6. Change Log

Each user has control over their personal details (name and password, covered earlier) via the Customer Management Portal plus:

| Date | Section | Change made |
|---|---|---|
| March 2021 | 3.1 | Added SIP over UDP content |
| ″ | 5.3 | Added Help Link content |
| ″ | 3.1, 4.2 | Added Site Caller ID content |
| ″ | 3.11 | Updated analogue line content |
| ″ | 3.15 | Updated migration content |