SIP Trunking Service Setup Guide

September 2024

Table of Contents

1 Introduction

This document is intended to assist BCM One customers and resellers in setting up their BCM One SIP Trunking service with an IP-PBX, a cutting-edge platform that has been tested and certified with BCM One's business-grade SIP Trunking service. For further assistance, please email us at support@bcmone.com.

2 SIP Trunking Service

BCM One's SIP Trunking service enable tested and certified IP-PBX platforms the ability to leverage customer-provided internet connection to make and receive phone calls.

2.1 Standard Service Plans Features

- Unlimited SIP trunk call paths
    - Broadcast and auto-dialer applications are considered separate and may have call-path restrictions
- Outbound dialing anywhere in the continental US and Canada
- Inbound calls on direct inward dialing numbers (DID)
- International calling capability
- Local E911 service
- Caller ID number
- Advanced user portal with real-time billing
- Multi-node redundancy

2.2 Advanced Features

- Caller ID name
- Inbound calling on toll-free numbers
- Outbound calling on international numbers
- Directory listings
- Multi-site local E911
- Remote call forwarding

3 Planning and Requirements

Before implementing your BCM One SIP Trunking service, it is important to address several key components.

3.1 Components of a Successful SIP Trunking Implementation

- Quality internet connectivity
- IP-PBX interoperability
- SIP-friendly network infrastructure
- IP-PBX configuration technical skills
- Specific BCM One/IP-PBX implementation guide
- Implementation worksheet
- Active BCM One SIP Trunking account
- Access to networking skills post-installation

3.2 Internet Connectivity Requirements

A reliable low-latency, low-packet-loss internet connection is required to enjoy the benefits of your BCM One SIP Trunking service, as shown below in Exhibit 3.2.

Internet Connectivity Specs for Desired Performance	
ISP Upload Speed	Enough Broadband for SIP Channels and Local LAN Traffic
ISP Latency	Less than 70 MS
ISP Packet Loss	Less than ½ percent
ISP Jitter	Less than 5 MS

Exhibit 3.2: Internet Connectivity Specs for Desired Performance

3.3 Bandwidth Utilization

3.3.1 Codecs

BCM One supports two codecs for inter-connectivity to and from the PSTN: G.711u and G.729a. If your network is not bandwidth constrained, you should be using G.711u. BCM One will recognize the codecs you set at the PBX level. For planning purposes, it is best to plan an 85k upload and download for each concurrent call using G.711u. However, plan to use an 35k upload and download for each concurrent call when using G.729a.

3.3.2 QoS/CoS

It is important to note when running SIP trunks over your own broadband connection using the public internet, there is no QoS. This is something to consider when sizing the appropriate internet connection to use for your SIP trunks and data traffic. Customers can address this by having enough broadband upload and download speed for their SIP trunking traffic and local-area network (LAN) traffic.

Customers can also address this by using a router with traffic-shaping/QoS features. In fact, many of today's routers have built-in voice-prioritization features. Customers who prefer to run two separate internet connections will segment off one connection for their LAN traffic along with a separate connection for their voice traffic. The use of two separate internet connections can give you a level of local redundancy not only from having two separate ISPs but also because you can use the BCM One auto-detecting disaster-recovery module to fail inbound calls to a separate internet connection in the event the primary connection is down.

When using the disaster-recovery module to fail calls to another ISP or internet connection, it is important to ensure your network is set up so that your PBX hardware can receive calls from either ISP. Section 8 of this implementation guide provides more information on the BCM One disaster-recovery module. For additional information on internet connectivity requirements or to request a free internet access quote, please email the BCM One sales team at sales@bcmone.com.

4 Service Set Up

In order to provide the highest level of service possible, BCM One uses an N+1 architectural model for call processing. You will need to ensure that your PBX and network-edge router/firewall are configured to accommodate this architecture.

4.1 Registration

- Registration is only required on the BCM One service if your PBX has a dynamic public IP address. If your public IP address is static, you do not need to register.
- If your public IP is dynamic, your registration timer should be no less than 120 seconds and no more than 300 seconds.
- If your public IP is static and you would like to register, although it is not required, your registration timer should be no less than 3600 seconds.
- If your PBX supports DNS SRV, your registrar IP is nexvortex.com
- If your PBX does not support DNS SRV, your primary registrar IP is PX11.nexvortex.com and your secondary registrar IP is PX15.nexvortex.com.
- Your authentication username and password were provided to you in the “new account email” you received when you first registered with our service.

4.2 Outbound Service

The most efficient way to ensure redundancy for outbound calling is to use DNS SRV for routing traffic to BCM One. If your PBX supports DNS SRV, pointing to nexvortex.com as your proxy IP address is all that should be necessary to ensure outbound redundancy.

If your PBX does not support DNS SRV, hopefully it supports configuration of multiple outbound proxies. If the system supports a primary and backup SIP proxy, use px11.nexvortex or px15.nexvortex.com for the primary SIP proxy, and use px11.nexvortex.com or px15.nexvortex.com for the backup SIP proxy. (You must use both px11.nexvortex.com and px15.nexvortex to be fully redundant.) Also, if the IP-PBX has a special address for SIP registrations or a registrar proxy, please change that to reg.nexvortex.com.

4.3 Inbound Service

As noted below in Section 5 of this document, you may receive SIP signaling from BCM One from any of the following IP addresses:

- 104.219.163.73
- 104.219.162.21

You must ensure that each of these IPs can pass UDP 5060 traffic into your network and that this traffic is port-forwarded, if necessary, to the internal IP of your PBX. Your PBX in turn should be configured with as many trunks as necessary to field traffic from these IPs. If you need additional assistance ensuring your local PBX configuration meets this requirement, please contact technical support.

5 Network Infrastructure

Network infrastructure is a critical component to your SIP implementation because routers, switches, and the firewall will need to be set up in conjunction with SIP signaling and RTP, as shown below in Exhibit 5-1.

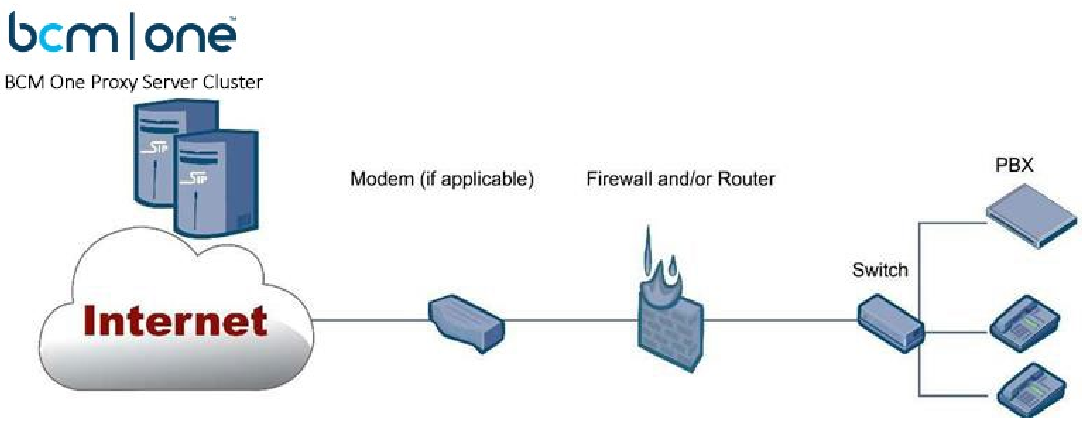

Exhibit 5-1: Network Infrastructure[1]

In your SIP signaling, you are specifying a great deal of information about how the call is to proceed. This information includes your IP address, the audio port you are using, dual-tone multi-frequency signaling (DTMF) configuration, etc.—all of this is controlled by how you configure your PBX. Your edge device (router and/or firewall) must be configured in concert with these PBX settings. Some edge devices can change the port ranges dynamically during calls as a security measure. This can cause dropped calls, loss of signaling, loss of two-way audio mid-call, or other problems if the firewall configuration is not correct for your particular PBX. When using an edge device—such as a firewall, network router, security application, etc.—support for the device or application is critical. BCM One does not configure your firewall, edge devices, security devices, etc. Generally, all edge devices should be configured to always allow UDP port 5060 traffic from the following IPs or subnets:

BCM One-generated graphic.

- 104.219.163.73
- 104.219.162.21
- 104.219.162.0/24
- 104.219.163.0/24

BCM One's SIP does not use session-border controllers (SBCs). For this reason, you must allow all IP addresses access to the UDP port range that your PBX uses for RTP. This UDP range depends on the configuration of your particular PBX and is configurable on most PBXs. Failure to do so may result in one-way audio issues.

6 SIP Service Specifications

SIP is a powerful protocol that enables the end user to control many aspects of service delivery. Although there are many ways to handle SIP, the information in Exhibit 6-1 below covers BCM One's IP trunking specifications and how your PBX should be configured to provide you with the best experience.

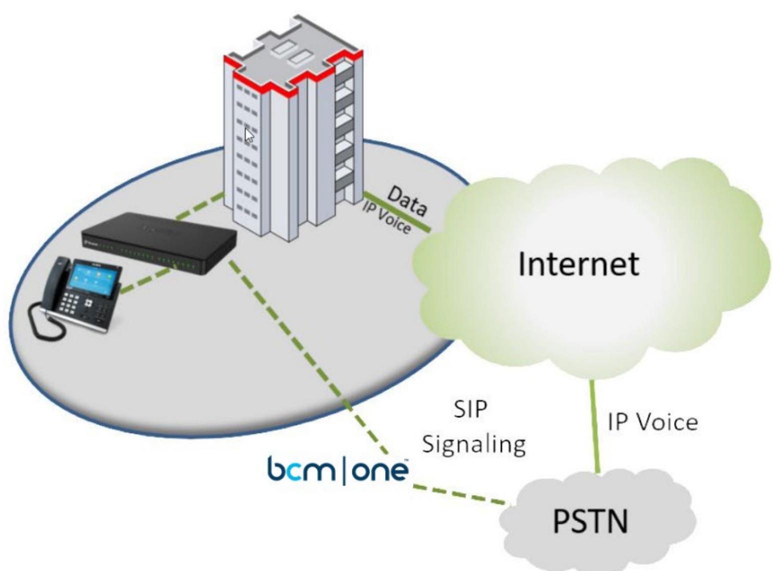

Exhibit 6-1: BCM One Service Logic SAM[2]

[2] Ibid.

6.1 Codecs

BCM One supports two main codecs: G.711u and G.729a. If your network is not bandwidth constrained, you should be using G.711u, not G.729a.

6.2 DTMF

BCM One officially supports RFC2833 as its only DTMF type. Your PBX must use RFC2833 to communicate DTMF to BCM One gateways. The DTMF type that you use to communicate between your PBX and your IP phones may be different. This will depend on the manufacturer of your IP-PBX and phones.

6.3 DTMF Payload

This should be 101.

6.4 Protocol

BCM One primarily supports SIP via UDP. If you would like to run BCM One service using SIP over TCP, let us know as early as possible, as our Network-Engineering team will need to make customizations.

6.5 NAT

BCM One does not perform ALG or SIP transformations on your traffic. Therefore, you must present your public IP address in the 'c=' portion of your SDP content.

6.6 Ports

UDP port 5060 is the SIP standard; it is the only port you should send communications to BCM One. You may use any local UDP port that you prefer; however, it must account for this in the configurations of your edge device (router and/or firewall) and PBX.

6.7 Fax

BCM One supports fax-over IP via G.711u pass-through and T.38. T.38 is the recommended format for reliability. BCM One provides this service as a best effort since fax over IP is extremely dependent on your IP connection and cannot be guaranteed.

7 Best Practices

7.1 Security

Passwords

It is essential that you change all default passwords to unique passwords, and extension passwords should never be the same as the extension number. When choosing passwords, ensure they are complex. If possible, require alphanumeric passwords with as many digits as the system allows.

Institute mechanisms to ensure employees change their passwords and access codes/PINS regularly. Moreover, delete former employees' passwords immediately following separation. Usernames and passwords should be erased when phones are repurposed/reassigned.

PBX

Consider limiting max trunk calls and max calls per extension in accordance with your company-specific requirements. When configuring your PBX, adhere to the following:

- Update your server's operating system and all associated software/firmware to the latest version and ensure all the latest security patches are enabled.
- Configure IVRs to use a timeout-based call-disconnect rule, as failure to do so could result in long calls.
- Disable remote notification, auto-attendant, call-forwarding and out-paging features if you do not use them.

Dial Plan Restrictions

It is important that you restrict your dial plan. If you do not make international calls, do not allow users to dial 011 as their first three digits. If you do make international calls, consider restricting allowable dial strings to only the country codes where you place calls.

Note: Do not forget to protect your dial plan against Caribbean dialing. You can access a list of country calling codes here.

Access

If your PBX supports access lists for IP authorization, they should be extremely conservative; moreover, limit VOIP registrations to office network or trusted networks.

If the PBX is configurable via a web browser GUI, it should not be accessible via a public IP. If you must make changes to your PBX configuration from outside your network, you should only enable remote access while you are working on the configuration and then immediately remove access when your updates are complete.

Additional Security

Please remember to enable VoIP logging to monitor activity and check firewall logs regularly to identify potential threats. In addition, assess the security of all PBX peripherals/applications regularly, including platform, operating system, password, and permissions scheme.
IP, unfortunately, is a high-value target for hackers. There are a few things you should do to ensure that your PBX installation is secure and well protected against the normal attack vectors for this technology.

8 Porting Numbers to BCM One

Processing a port order with BCM One is an easy process. To get started, you'll need to collect a few pieces of information that will simplify the process. Before processing a port order, you should have the following information:

First, you need your account information as it appears by your current carrier, including your business name, the service address(es) for the number(s), your billing telephone number (BTN), and the name of the person authorized to make decisions on your account.

Note: the authorized person is the one who must sign the letter of authorization (LOA.)

Second, you will need a copy of your current carrier invoice, meaning less than 60 days old. If you are porting a toll-free number, the invoice should have the toll-free number listed.

Third, you need a detailed list of which numbers you want to port. If you are porting all your telephone numbers, this is considered a full port. (We will ask you this question during the order-entry process.) If you are not porting all of the numbers, then this is referred to as a partial port.

Finally, you will need the above mentioned criteria for each location or carrier account that you will port numbers from. If you have any questions, please email us at postsales@bcmone.com or call us at 855-639-6300.

The port-order process is conducted in the following five steps:

1. Order type, including main contact information
2. Site addresses
3. Billing telephone numbers
4. Phone numbers building the list
5. Documents (LOA, carrier invoices, etc.)

8.1 Step 1 Port-Order Type and Contact Information

In this step, you need to identify the type of numbers you want to port. Telephone numbers and toll-free numbers require separate orders. Choose telephone numbers or toll-free numbers. Enter the name, phone number, and email address of the primary contact for this port order, as shown below in Exhibit 8.1.

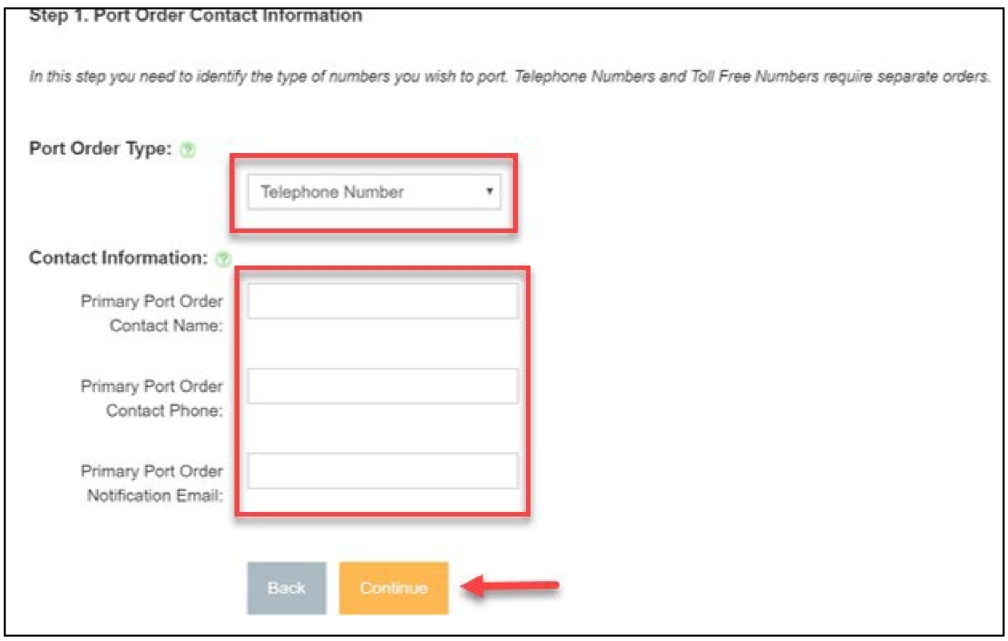

Exhibit 8.1: Port Order Contact Information[3]

Click Continue when complete.

[3] Ibid.

8.2 Step 2: Locations

The addresses listed for your locations must match the addresses on the telephone bill the numbers are being ported from, as the carrier may reject the request.

In the screenshot below in Exhibit 8.2, add addresses for all locations you are porting. Click on the + button and complete the dialog box to enter the address for a location. When you click the Add button, the location will appear in the list of locations at the top of the screen.

If only one location is needed, the user can proceed to the next step by clicking on the Add button.

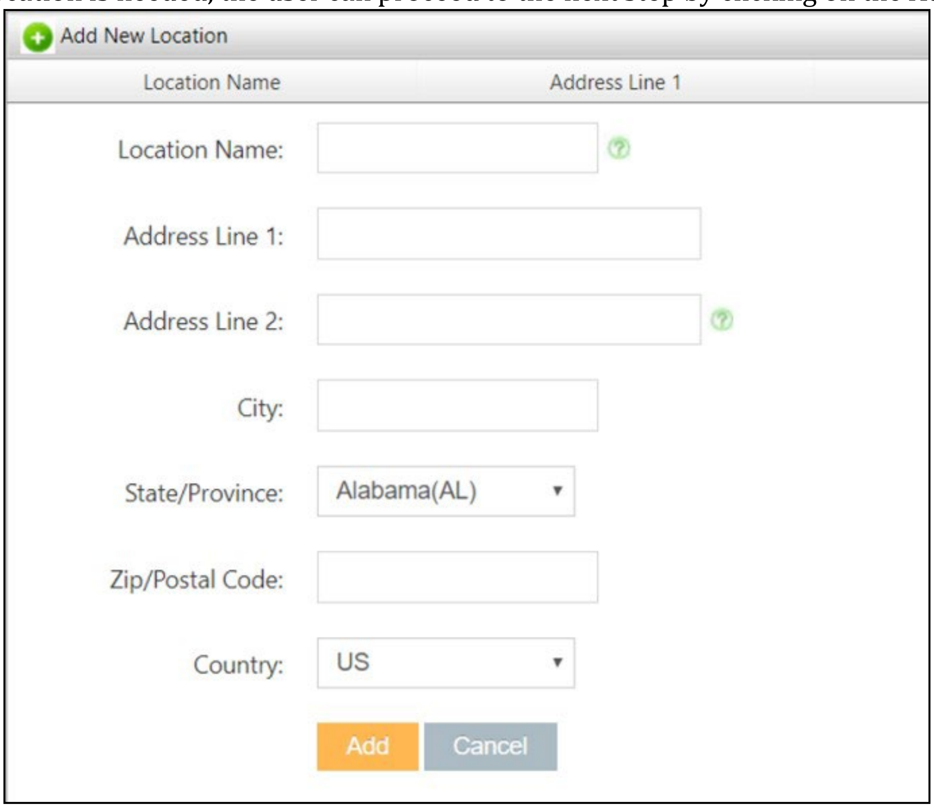

Exhibit 8.2: Port-Order Address Tab[4]

8.3 Step 3: Billing Telephone Numbers

Telephone companies identify accounts based on a billing telephone number (BTN), which should appear near the address at the top of the bill. You will need to add this number for each location and

[4] Ibid.

carrier. In addition, you must select if you want to port all the numbers from this account, which is considered a full port type. (Note: a partial port type would only port some of the numbers.) If you are unsure, it is probably best to contact your carrier and ask which BTN belongs to which number(s). When you enter the correct BTNs, please click on the Continue button, as shown below in Exhibit 8.3.

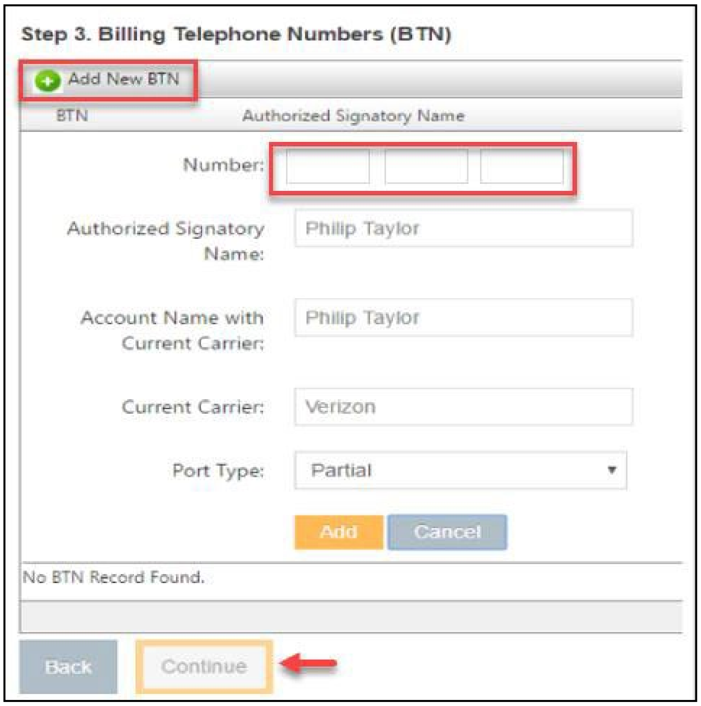

Exhibit 8.3: Billing Telephone Numbers Drop-Down Menu[5]

8.4 Step 4: Adding Telephone Numbers to Be Ported

In this step, you will upload the telephone numbers you want to port. Please ensure that the numbers are attached to the correct location and BTN. There are two ways of doing this regarding the numbers you want to port. If there are only a few numbers, you can enter them individually; if they are sequential, you can enter a range. If the numbers are numerous but not sequential, you can use the free-form comma separated input.

Please ensure the numbers entered are the ones attached to that location and BTN. Once entered, the numbers will appear on the list below the form. If you would like the Calling Name ID to be transmitted with calls from certain numbers, indicate that in the form, as shown below in Exhibit 8.4.

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[5] Ibid.

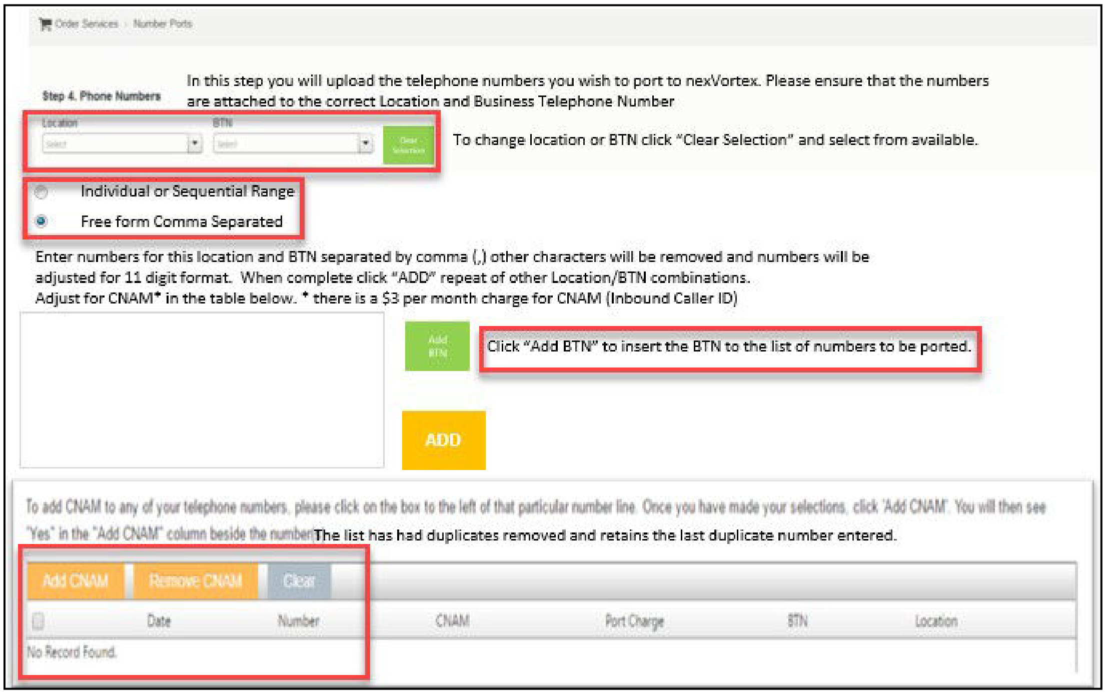

Exhibit 8.4: Adding Telephone Number to Be Ported Drop-Down Menu[6]

8.5 Letter of Authorization (LOA) Section

In order to allow us to work on your behalf to port your telephone numbers, please do the following:

First, you need to give us your official authorization to work on your behalf through a letter of authorization, which you can obtain by clicking on the Generate LOA button below in Exhibit 8.5.

Second, print the portable document format (PDF) file and sign and date it and then scan the document and click Upload LOA, which allows you to select the file and upload it, as shown below in Exhibit 8.5. Note: If you have multiple locations and/or BTNs, there will be a separate LOA for each. The person authorized must sign and upload them for each account. If there are numbers that will not be ported, you can put these in the Notes or Special Instructions box.

[6] Ibid.

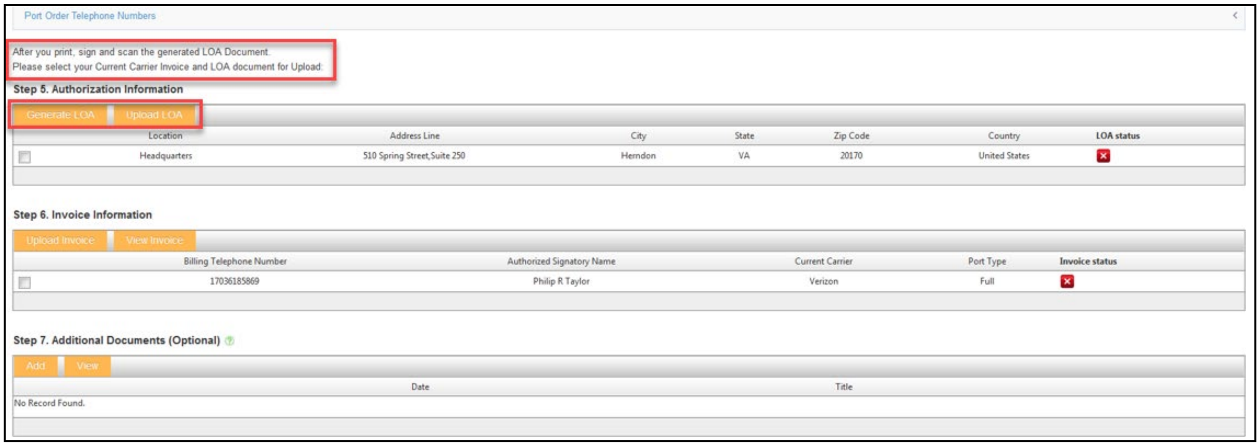

Exhibit 8.5: Port Order Telephone Numbers Drop-Down Menu[7]

The Telephone Numbers drop-down menu is also where you send us the current copy of your carrier invoice. In fact, we can answer most questions that arise in the process when we have your carrier invoice. If you have multiple carriers or invoices from different locations, please upload them as well. Simply click on Upload Invoice and select the correct file(s), as shown in Exhibit 8.5.1 below. Again, we will need one for each location or carrier account.

If you have additional information or wish to provide additional information, upload documents in Step 7 in Exhibit 8.5.1 or add directions into the textbox in section 8. When you have green status in Steps 5 and 6 and have provided additional information, if applicable, click on the Submit button and this will begin the process. BCM One will contact you regarding additional information and preferred porting dates.

[7] Ibid.

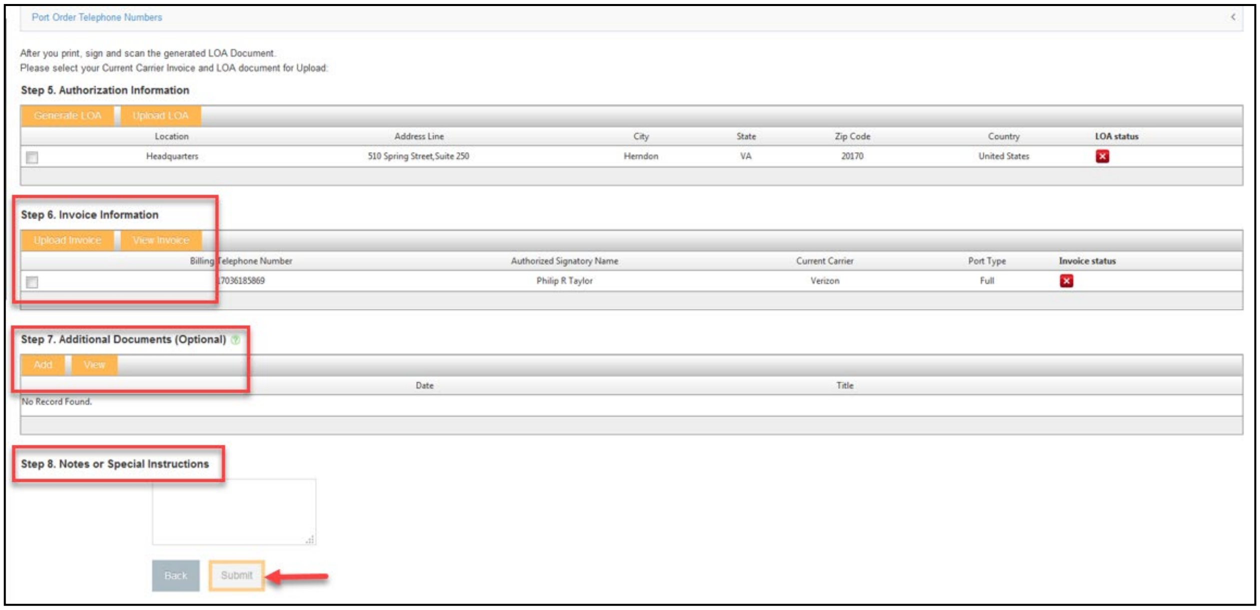

Exhibit 8.5.1: Port Order Invoice Drop-Down Menu[8]

9 E911 Implementation

At BCM One we want you to enjoy your services; therefore, we strive to make our products as seamless as possible for the customer. To ensure our customers receive top-notch service, we periodically remind our clients to check their 911 settings. We recommend this because using voice over internet protocol (VoIP) services is different than a traditional phone service since the physical locations can be hard to locate, as IP networks are distributing the calls.

E911 is important to the safety and well-being of your employees, but unlike a traditional phone service, you need to manage your E911 service locations, as it has implications beyond the safety of your personnel. Some states have been amending their laws to set new 911 requirements; in fact, more companies have remote offices or employees using company-issued VoIP phones at home. This can lead to legal ramifications or liability exposures. You cannot assume in emergency situations people will know which phones to use, so you must ensure all company telephones are configured correctly in order for first responders to quickly arrive at the correct location.

We have instituted best practices to make this process as easy as possible, thereby ensuring backup options in the event of disasters that might interfere with data communication.

[8] Ibid.

Below is a walkthrough of how to check your current E911 settings and make corrections. When you have made changes or are verifying your current settings, do not dial 911. There can be serious consequences when calling 911 for testing purposes. Please dial 933 to verify the address for each location.

There are a few components required to create a valid E911 entry, and to ensure everything is accurate, you will need to verify the following items: For E911 to work, there needs to be an address, which is verified and registered, and a registered phone number (calling line ID). In addition, the system can use the proxy key and the IP address (if it is static and public) as backup methods to identify callers.

9.1 Add an Address into the System

To add an address, you need to access the user portal at https://cloudcom.nexvortex.com/Login.aspx and go to MyAccount> MyAccount Settings where the addresses associated with the account are listed. Please ensure any addresses that will be used for 911 are validated; this means our 911 supplier have vetted them as matching its national database and the appropriate answer point is known. Addresses currently assigned to 911 will have a check mark in the 911 column.

If you need to add another address, click the + button and fill in the form, as shown below in Exhibit 9.1. When you click the Order button, the system will verify that this is a valid address or provide you with one that has the more detailed zip code.

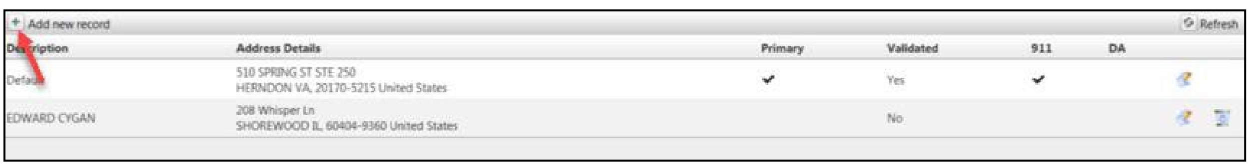

Exhibit 9.1: The Add New Record Tab[9]

9.2 E911 Address

Now that the address is in the system, the next thing you need to do is assign it to the 911 address. While you are still in the portal, select Settings > E911 Locations. Any address that had a check mark under 911 in the previous table should be in this table. To add a new location, click on the icon at the right-most column, as shown below in Exhibit 9.2. If the contact name is different for this location, enter it in the Contact Name box.

[9] Ibid.

Use the pull-down tab next to Address to select from the entered addresses, and then use the pull-down tab to select the phone number that will be the primary number for this location. If you are not sure what the actual calling line identification (CLID) for this location is, contact your PBX administrator, IT department, or contact us at support@bcmone.com.

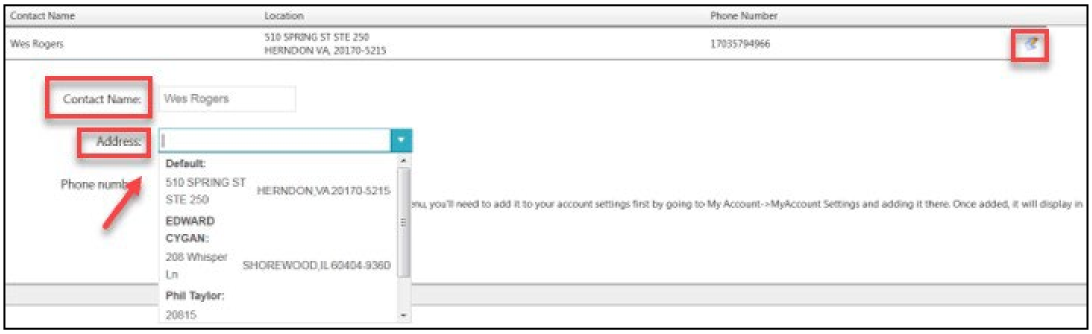

Exhibit 9.2: Assigning an Address Menu[10]

When you are confident that you have correctly entered the information, click on the Order button. Each service plan includes a single 911 location and each additional location (including a temp DID) is $2.50 ($4.95 for hosted).

Mapping to Additional Information

The last step involves connecting phone numbers, proxy addresses, or IP addresses so that if there is corruption within the caller ID, we can determine where the call was sent from. Select Settings >E911 Links. Each 911 location will appear in its own box. Click on a location and the dark green border will turn a bright aqua color. Click on the green Show Configured Items button and all phone numbers, proxies, or IPs for this location will appear, as shown in Exhibit 9.2.1.

If you have additional numbers attached to this address, select Phone Based from the Step 2 pull down and a list of numbers associated with the account will appear, as shown below in Exhibit 9.2.2. Carefully choose and click on the telephone number(s) that might be transmitted from this location. They will appear in the location box. If you know the proxy ID for this location, you can also use the pull down to select Proxy or IPAddress based and a list of proxy values or fixed IP addresses will appear, as shown below in Exhibit 9.2.3. Select the correct proxy value for this site. If the site has a fixed public IP, it should also appear in the list and associated with the address. As you click on each tab, the address will appear in the aqua box. Repeat this process for each registered location.

Note: If you are not sure of any of this information, do not add it. This may result in a 911 call going to the wrong emergency services answer point. If in doubt, contact your PBX administrator, IT department, or contact us at support@bcmone.com.

[10] Ibid.

Exhibit 9.2.1: Step 1 Please Select Location Icon[11]

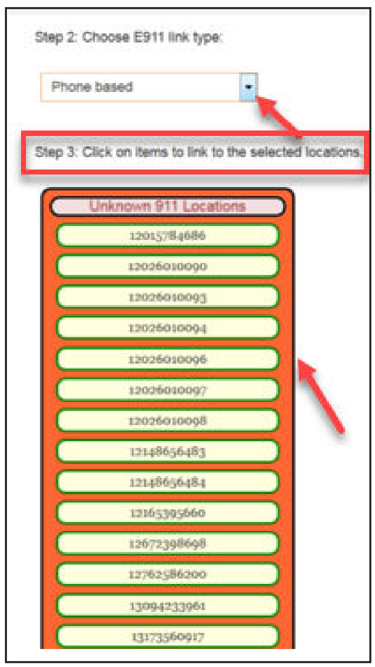

Exhibit 9.2.2: Step 2 Choose E911 Link Type Phone Based Icon[12]

[11] Ibid.
[12] Ibid.

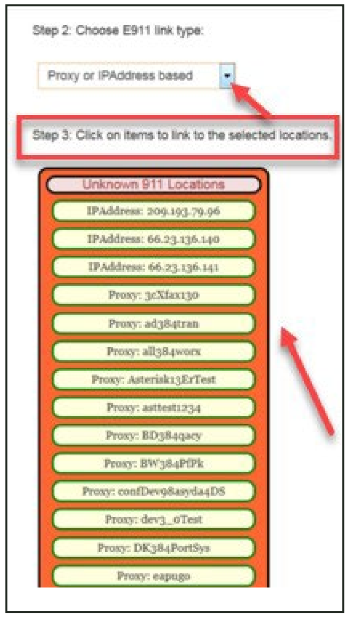

Exhibit 9.2.3: Step 2 Choose E911 Link Type Proxy or IPAddress Based Icon[13]

To associate number, proxy, and IP with this address, select one of the applicable orange boxes from the drop-down menu and the address will appear in the address box. If you select an address and click on the green Show Configured Items button, all items assigned to this address will appear, as shown below in Exhibit 9.2.4. To remove an item, simply click on it.

Exhibit 9.2.4: Show Configured Items Icon[14]

[13] Ibid.
[14] Ibid.

9.3 Verify Submitted Information

When you have completed making additions, follow the same procedure for all other addresses and click on the green Review 911 Changes box to review your changes, as shown below in Exhibit 9.3. If an item is incorrect, click on the X at the right-most column and it will be removed, as shown below in Exhibit 9.3.1.

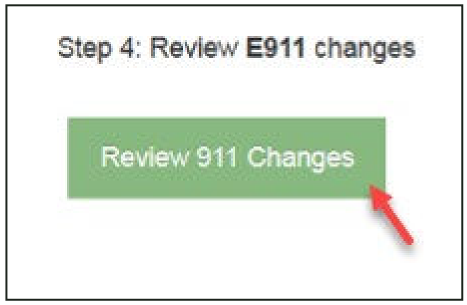

Exhibit 9.3: Review 911 Changes Box[15]

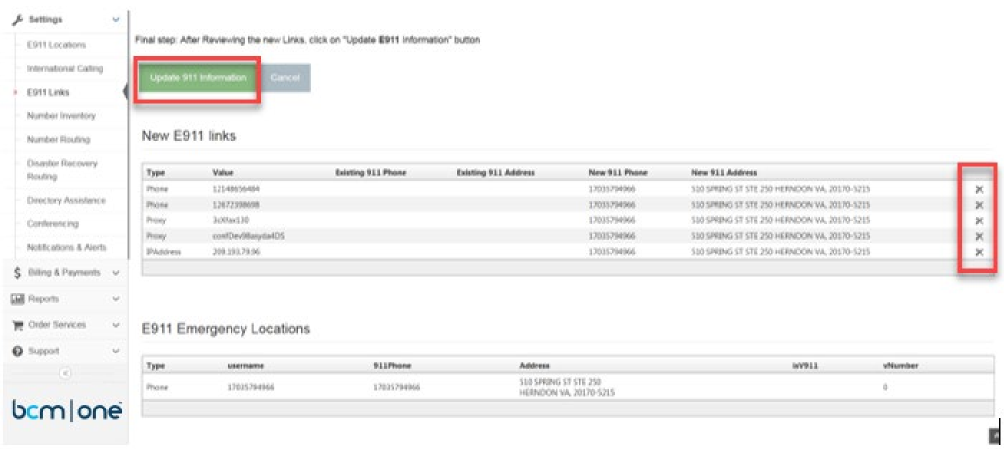

Exhibit 9.3.1: Illustration of the Update E911 Information[16]

When you are satisfied with your changes, click the green Update 911 Information button. You have now completed your updates.

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[15] Ibid.
[16] Ibid.

9.4 911 Testing

From each location, dial 933 and listen as the recorded message reads the E911 address associated with that phone number to verify the information they are providing is their address.

We will periodically remind you to check your settings for E911 and other important account settings. We do recommend that you check everything at least twice a year, more frequently if you have any location changes or many remote users.

10 Multi-Site E911

For multi-site E911 the first step is to register each physical site's address in your account web portal. This is done by clicking on Order Services> E911> Add, as shown below in Exhibit 10-1.

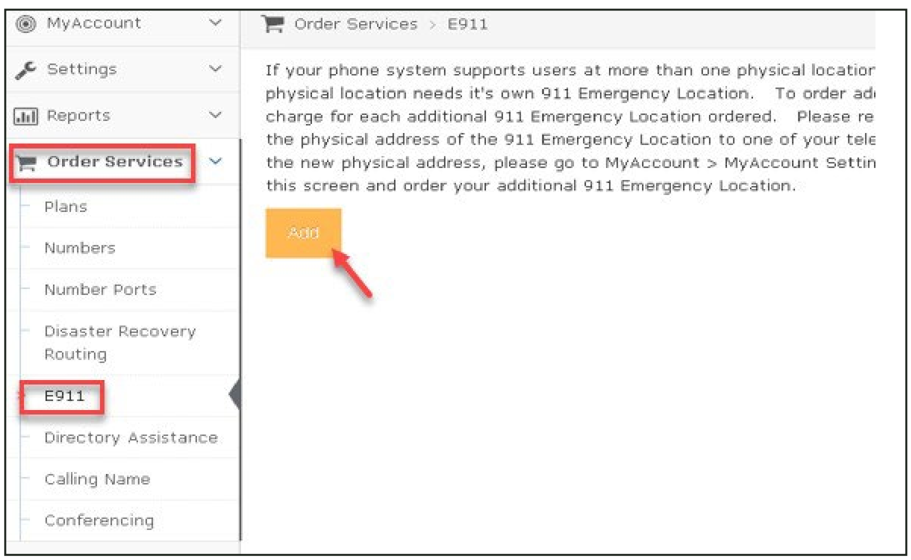

Exhibit 10-1: Multi-Site E911 Setup Menu[17]

The second step is to open a ticket within BCM One support so that a BCM One team member can review your multi-site E911configuration. Your multi-site E911 configuration may be different, depending on if you are using an IP-PBX at each location or if you are using remote phones. To open a ticket in BCM One support, please email support@bcmone.com**.** It is best to list your account number in the subject line along with the words multi-site E911. Also, if you need to add additional E911 emergency locations to your account or delete an existing 911 emergency location, contact us

[17] Ibid.

at support@bcmone.com. There is additional information on E911 and multi-site E911 in your web portal under Support > Set Up Guides.

11 Number Routing

To set your inbound number routes you will need to log in to your web portal and select Settings > Number Routing. You will load your DID numbers and toll-free numbers, if applicable, in this section of your portal. The number-routing feature in your web portal enables you to add or change your inbound call routes. You can also change the way the call is presented to your PBX equipment. To change or add a number route for one of your DID numbers or toll-free numbers, select a number from your routing table and select edit. You will see a pop-up window that will let you set up your inbound routes and preferences for that given number. The changes you make are in real time. You can choose to use a static IP (preferred) or a dynamic IP set up, as shown below in Exhibit 11.

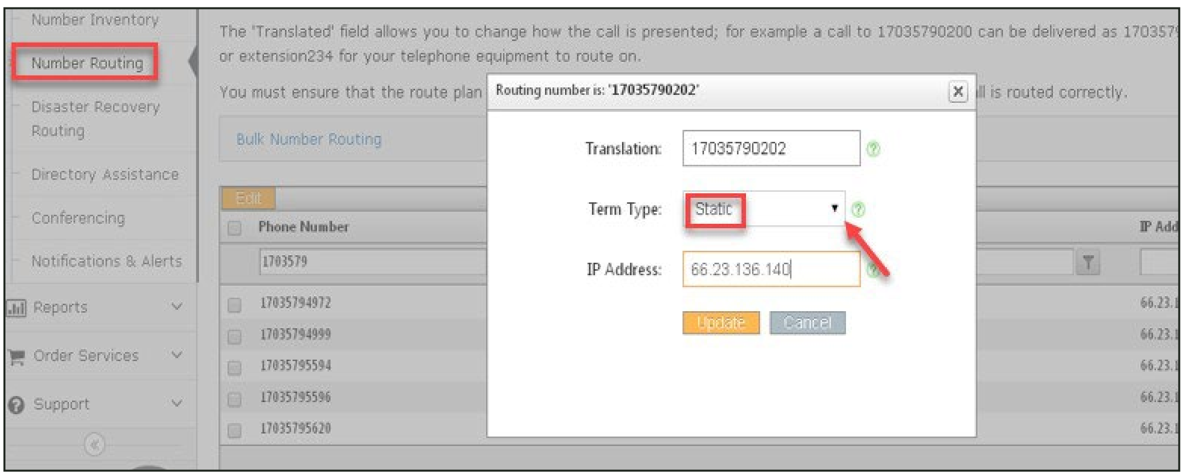

Exhibit 11: Choosing a Static or Dynamic IP Pop-Up Window [18]

11.1 Static IP for Inbound Number Routing

With the term type set to static, many variations can be entered into the IP Address/Contact field. Some commonly used entries are Static IP (e.g., 104.219.163.73), Dynamic DNS (e.g., nexvortex.dyndns.com), DNS (e.g., sip.nexvortex.com). It is important that this field corresponds to the LAN where the phone system is installed. The translation field must match a configured inbound route on the end-user PBX. What is entered in this field will be populated as the USER part of SIP URI, also known as the request URI, of the invite delivered to the end-user PBX for an

[18] Ibid.

inbound call. In order to support standard North American numbering as well as E.164 numbering formats (a plus + sign will appear in the sip_uri of the Request URI, To, and From headers), whenever possible, the inbound routes that you configure on your PBX should use wildcarding for DID recognition. For example, if your DID is 17035790200, you will want to configure your inbound route on your PBX as *7035790200. This will match both 17035790200 and +17035790200.

11.2 Dynamic IP for Inbound Number Routing

Based on the end user's registration message FROM header, whatever is presented as the USER part of the header field is what the customer needs to enter as both translation and IP Address/Contact. This is typically the username BCM One provided with your new account or the default phone number on your account.

11.3 Inbound DID Registration

If your public IP is dynamic, you will need to register with BCM One to receive inbound traffic. The registrar IP you configure in your PBX should be reg.nexvortex.com **a**nd your timer expiry for this registration should be no less than 300 seconds. If your public IP is static, you should not register at all, as this information is completely unused for your service delivery with BCM One.

12 Disaster-Recovery Routing

Disaster-recovery routing with automatic detection allows incoming calls to be forwarded to an alternate destination in the event the call(s) could not be delivered to the intended location due to a problem with your phone system or your internet access connection. Each number is individually configurable, and you can specify up to three alternate locations. Each alternate location can be either a standard telephone number or an IP end point. For example, you can forward calls to your cell phone, analog lines, PRI number, or a different internet connection. (Note: Calls routed back out to the traditional phone network will be treated as normal outbound calls for billing purposes.) To set up disaster recovery, you need to log in to your BCM One web portal and then click on Settings > Disaster Recovery Routing, as shown below in Exhibit 12.

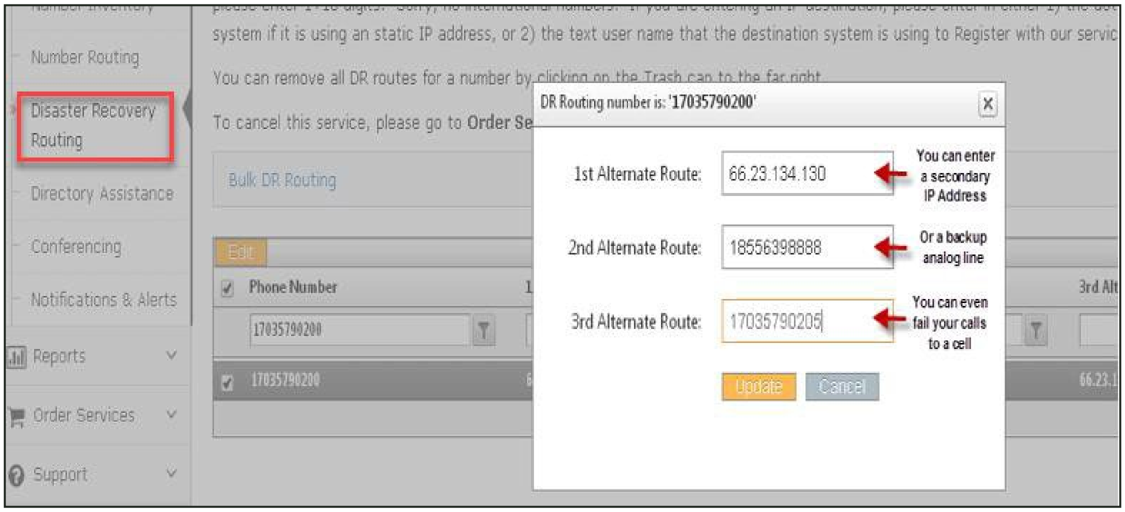

Exhibit 12: Choosing a Static or Dynamic IP Pop-Up Window

In Exhibit 12, you can see the different routes you can set your numbers to. Using the tabs, you can set up your numbers to fail over to another number, such as a cell phone or another land line.

13 Troubleshooting

Please review the following troubleshooting steps:

13.1 Check to See if Your Service Is Enabled

In your account portal at the top of your screen you will see a real-time status bar. Your account should be in an Active Status to use the service.

13.2 Customer System – Outbound Call Failure

Check for the following:

- The system is pointing at nexvortex.com.
- Port 5060 is open on the firewall.
- NAT translation is correct between LAN private IP address and public IP address.
- You have the correct proxy username and password configured.

If you are using 7-digit dialing, ensure that the dial plan in your PBX is configured to prepend 1+ area code before the 7-digit number since all domestic calls presented to BCM One must be 1+10 digits or 10 digits.

13.3 Customer System – Inbound Call Failure

Some systems require nexvortex.com for verification to be

configured. Check for the following:

- Port 5060 is open on the firewall.
- NAT translation is correct between LAN private IP address and public IP address.
- You have set up the IP route for the number correctly with BCM One. This is done through the customer or reseller Partner Connect portal by selecting Settings > Number Routing.
- The dial plan is configured to route the number to a valid location on your customer system.

13.4 One-Way Audio or No Audio after Call Is Set Up

Check that the RTP audio ports are open on the firewall. Confirm that NAT translation is being handled correctly and that public IP addresses are being sent in the SDP data of invite messages sent to BCM One.