UNIVERGE® SV9300

UNIVERGE BLUE® CONNECT BRIDGE Setup Guide

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NEC UNIVERGE BLUE CONNECT BRIDGE

SECTION 1 NEC SV9300 AND NEC UNIVERGE BLUE CONNECT BRIDGE SETUP GUIDE

1.1 This Guide and Related Documents

This guide was created to assist knowledgeable vendors with configuring the NEC SV9300 Communication Server with NEC UNIVERGE BLUE CONNECT BRIDGE. It provides sample entries for the required fields. The actual data is provided by NEC UNIVERGE BLUE CONNECT BRIDGE when service is activated. Questions about software and hardware installation or other PBX configuration issues should be directed to NEC's National Technical Assistance Center (NTAC).

For complete details on using SIP trunks with the SV9300, refer to the SV9300 Networking Manual.

For complete details on using DID features, refer to the DID feature in the SV9300 Features and Specifications Manual.

For details about related hardware, refer to the SV9300 System Hardware Manual.

These manuals can be downloaded from NEC's National Technical Assistance Center (NTAC) web site. You must have a valid dealer ID to access the documents.

1.2 NEC UNIVERGE BLUE CONNECT BRIDGE Account

1.2.1 Overview

This document outlines the setup process for a UNIVERGE BLUE CONNECT BRIDGE account, with specific focus on the steps required to integrate BLUE CONNECT BRIDGE with the on prem NEC PBX. This includes gathering all necessary information before beginning the setup process, creating the UNIVERGE BLUE account, creating Trunk Users and creating a Tie Trunk.

1.2.2 What this Document does not Address

This document is not a comprehensive UNIVERGE BLUE CONNECT BRIDGE setup guide and does not cover all possible account setup steps. Beyond the steps required to integrate with Bridge, this document does not contain information on how to setup an on-premise NEC PBX. It does not contain or address local networking or firewall setup.

Table 1 Defined Terms

Term	Description

CONNECT BRIDGE	CONNECT BRIDGE is a service subscription designed to extend and cloud-enable the on-prem PBX with cloud-based Team Collaboration, Meeting, File sharing, and Mobility:
	❍ UB CONNECT Team Chat
	❍ UB Meet
	❍ UB Share
	❍ Presence
	❍ Calling: feature set varies based on customer's decision to integrate BRIDGE with the on-prem PBX
	    ❍ With PBX Integration: Ext to Ext AND PSTN dialing from both NEC Deskphone and CONNECT BRIDGE apps
	    ❍ Without PBX integration: Ext to Ext dialing only
	❍ Access to all Chat, Meet, Share, Calling capabilities from within UB CONNECT desktop and mobile apps

PBX Integration	The name of the feature allowing integration between a UB **CONNECT BRIDGE** account and an on-premise PBX. Integration between the 2 platforms is accomplished via the use of a **Tie Trunk**.
Tie Trunk	A virtual SIP Trunk connecting one or more on-premise PBX with a UB **CONNECT BRIDGE** account for the purpose of routing phone calls from the PBX to **Trunk Users** on the UB **CONNECT BRIDGE** account or vice versa.
CONNECT BRIDGE User	A descriptive term used to describe a User created on a UB **CONNECT BRIDGE** account and has a CONNECT BRIDGE license assigned. CONNECT BRIDGE Users cannot dial PSTN phone numbers until they are configured as a Trunk User.

Table 1 Defined Terms (Continued)

PBX User	A descriptive term used to describe a User that exists in the on-premise PBX. PBX user has a terminal and services provided by the on-premise PBX.
Tie Trunk User	A User who has both an extension in the on-premise PBX (and hence a terminal) and a UB **CONNECT BRIDGE** User/Ext for the purpose of using UB **CONNECT BRIDGE** services and apps. The **Trunk User** setup includes creating the same extension on both platforms. **Trunk Users** are created by creating both a **CONNECT BRIDGE User**, and a **PBX User**, then configuring both sides to route appropriately via the **Tie Trunk**. **Trunk Users** can dial PSTN phone numbers.
User's Extension	The extension number assigned to the User on both the on-prem PBX, as well as the **CONNECT BRIDGE User** in the UB Control Panel. The user themselves sees this extension on their NEC terminal, as well as on their CONNECT BRIDGE apps.
Premise Extension	A descriptive term used within the UB Control Panel to describe extensions that exist in the on-premise PBX but which are not tied to a BRIDGE User (Ex. Common Area phones, other users not using BRIDGE, etc). The UB **CONNECT BRIDGE** account uses this knowledge to understand when to route a call to a specific extension to the on-premise PBX over the **Tie Trunk**.
Cloud Extension	A descriptive term used within the UB Control Panel to describe a set of hidden extensions that are used by the on-premise PBX to route calls over the **Tie Trunk** to the UB **CONNECT BRIDGE** account to ring the **Trunk User's** desktop and mobile applications. A **Cloud Extension** is required when assigning a User to a **Tie Trunk**. The **Cloud Extension** is arbitrary and can be any extension not currently in use. Best practice is to choose a similar extension in another number range. (Ex. For **User Extension** 1001, choose a **Cloud Extension** such as 5001, etc.)

1.3 Deployment Planning

Before beginning the setup process, it is recommended to plan out your installation in advance and gather the necessary information you will need to setup the Univerge Blue account in the Control Panel.

The following information should be gathered for each Trunk User that will be created on the Univerge Blue account:

User's Name	User's Email	User's Extension	Cloud Extension	Voicemail PIN
E.g. John Doe	E.g. johnd@domain.com	E.g. 1001	E.g. 5001	E.g. 123987

IMPORTANT: Cloud Extensions are any, arbitrarily selected extensions, which are free on both Univerge Blue CONNECT BRIDGE and NEC PBX. i.e., if your NEC PBX is using 1xxx extension range, use the 5xxx extension range for the Cloud Extension.

In addition to User Information, the following information should be gathered to assist in setting up the Tie Trunk:

Item	Answer
NEC PBX model	
NEC PBX PSTN trunk access code	
NEC PBX extension length (Lengths 3, 4 and 5 are supported)	
Tie Trunk Codec priority (G.729 and G.711 are supported)	
NEC PBX hardware keycode	
Number of concurrent phone calls needed on the Tie Trunk (This is determined by the number of concurrent call licenses on the on-prem PBX)	
Complete list of Premise Extensions	
PBX WAN IP address	

Once all information is available, proceed to the account setup.

1.4 UNIVERGE BLUE CONNECT BRIDGE Account Setup

1.4.1 High Level Steps

The required steps for creating and setting up a UNIVERGE BLUE CONNECT BRIDGE account are:

1. The creation of a quote.
2. The addition of a CONNECT BRIDGE license to the quote.
3. Finalizing the quote and creating the account from that quote.

These steps will ensure a CONNECT BRIDGE account is created with the required features and functionality.

1.4.2 Creating a Quote

Quotes are created in the Quote Tool located within the Partner Portal.

1. Once logged into the Partner Portal, from the home page, click on the quick link for **Quotes**.

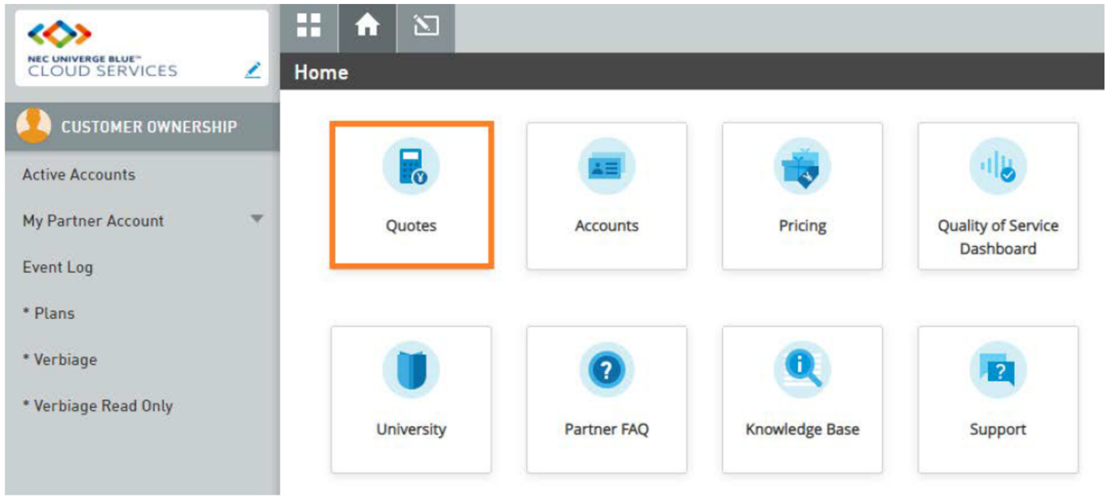

Figure 1 Home Page – Quotes Link

2. Next, click on the button to **Add new quote**.

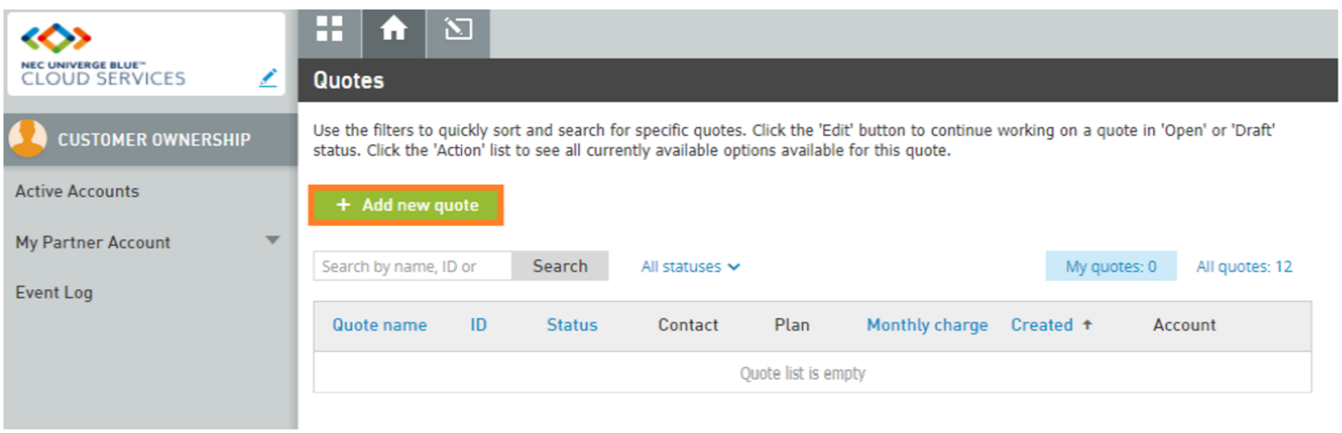

Figure 2 Quotes – Add New Quote

3. Provide a name for the quote and click **Create quote**.

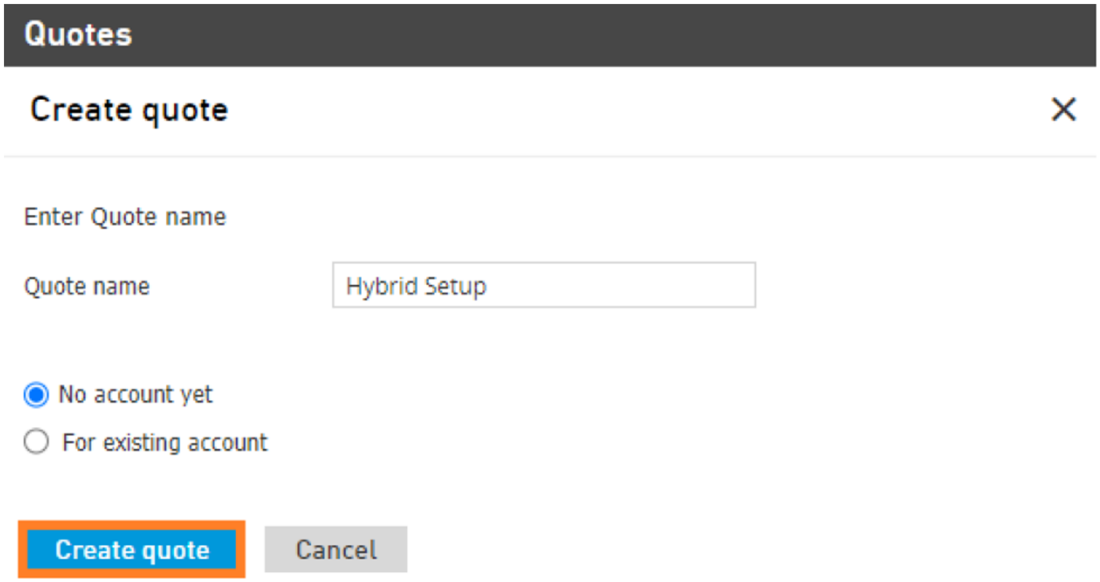

Figure 3 Quotes – Create Quote

4. Provide the address for the account's main location. Optionally, add additional locations to the quote. When finished adding locations, click **Next**.

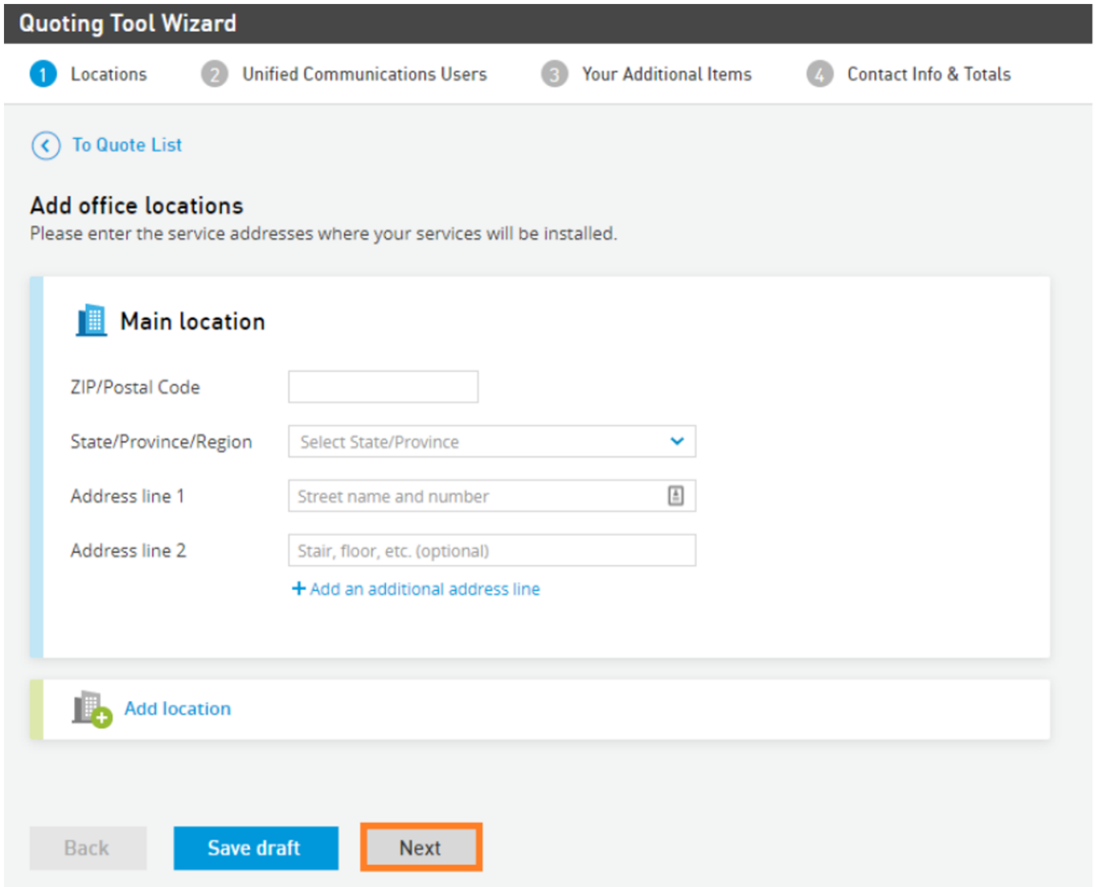

Figure 4 Add Office Locations

5. At a minimum, select one of the following licenses and add it to the quote then click **Next**.
   - CONNECT BRIDGE (NEC-J Partners will only see one license called **Connect**)
   - CONNECT BRIDGE ProPLus

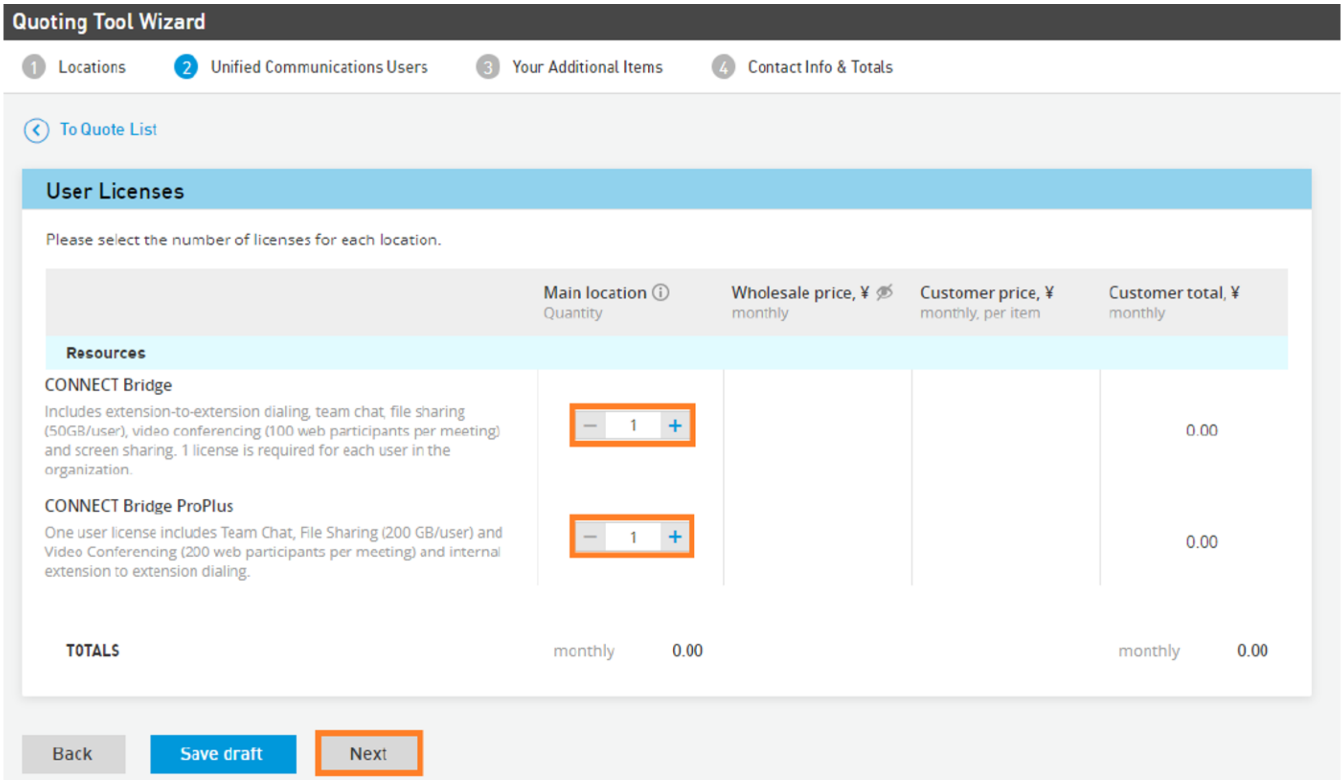

Figure 5 User Licenses

6. Complete all remaining sections of the quote. When complete, click the **Save opportunity** button. The quote will appear in your list of quotes. To create the account, click the **Create account** button on the quote. This action will launch the Create Account Wizard.

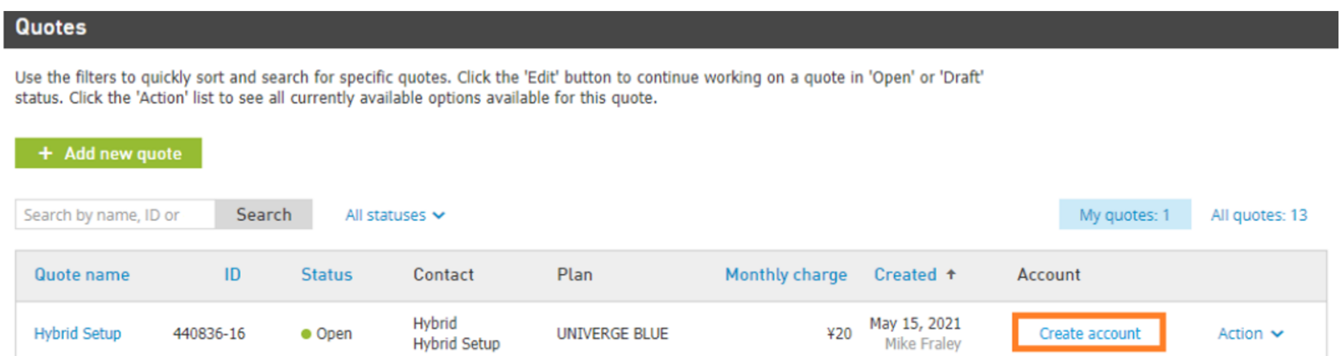

Figure 6 Create Account

1.4.3 Create Account Wizard

The Create Account Wizard (CAW) is a short series of steps that gathers the minimum information required to create an account in the Control Panel.

1. First, provide the account's access information. The information entered on this page will be used by the Account Owner when accessing this account in the Control Panel.

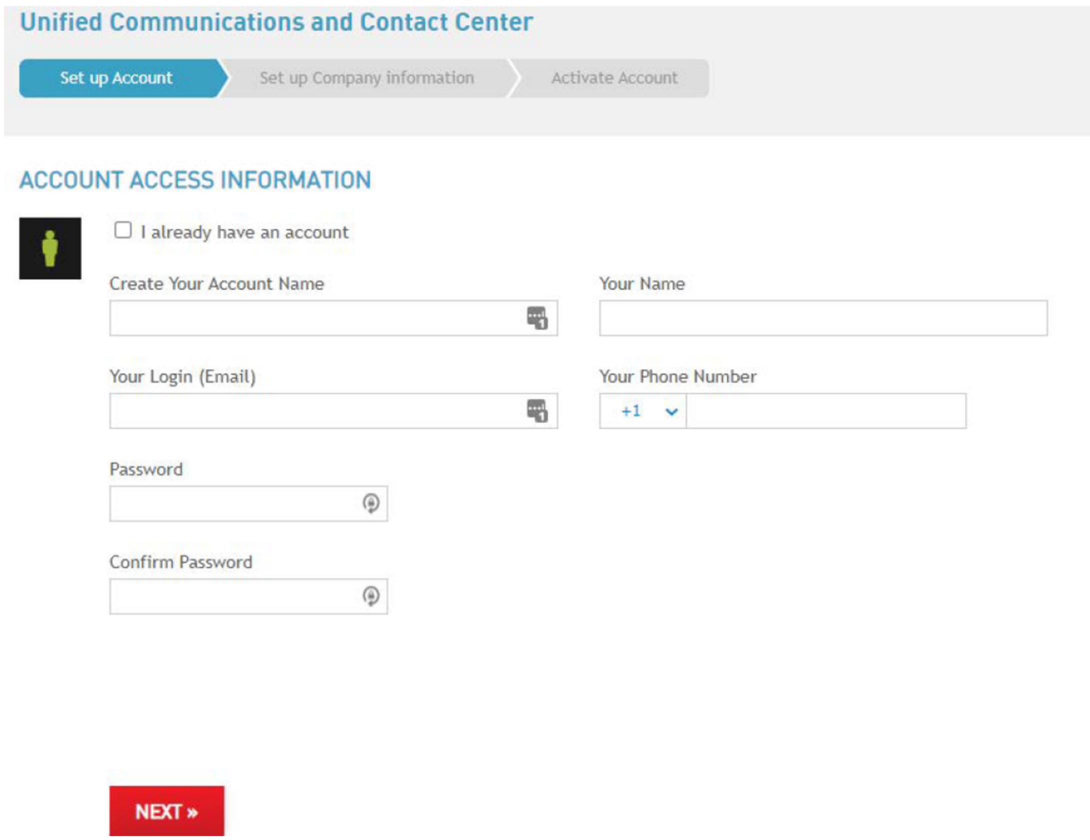

Figure 7 Create Account Wizard

2. Next, provide the company's main address, language and time zone.

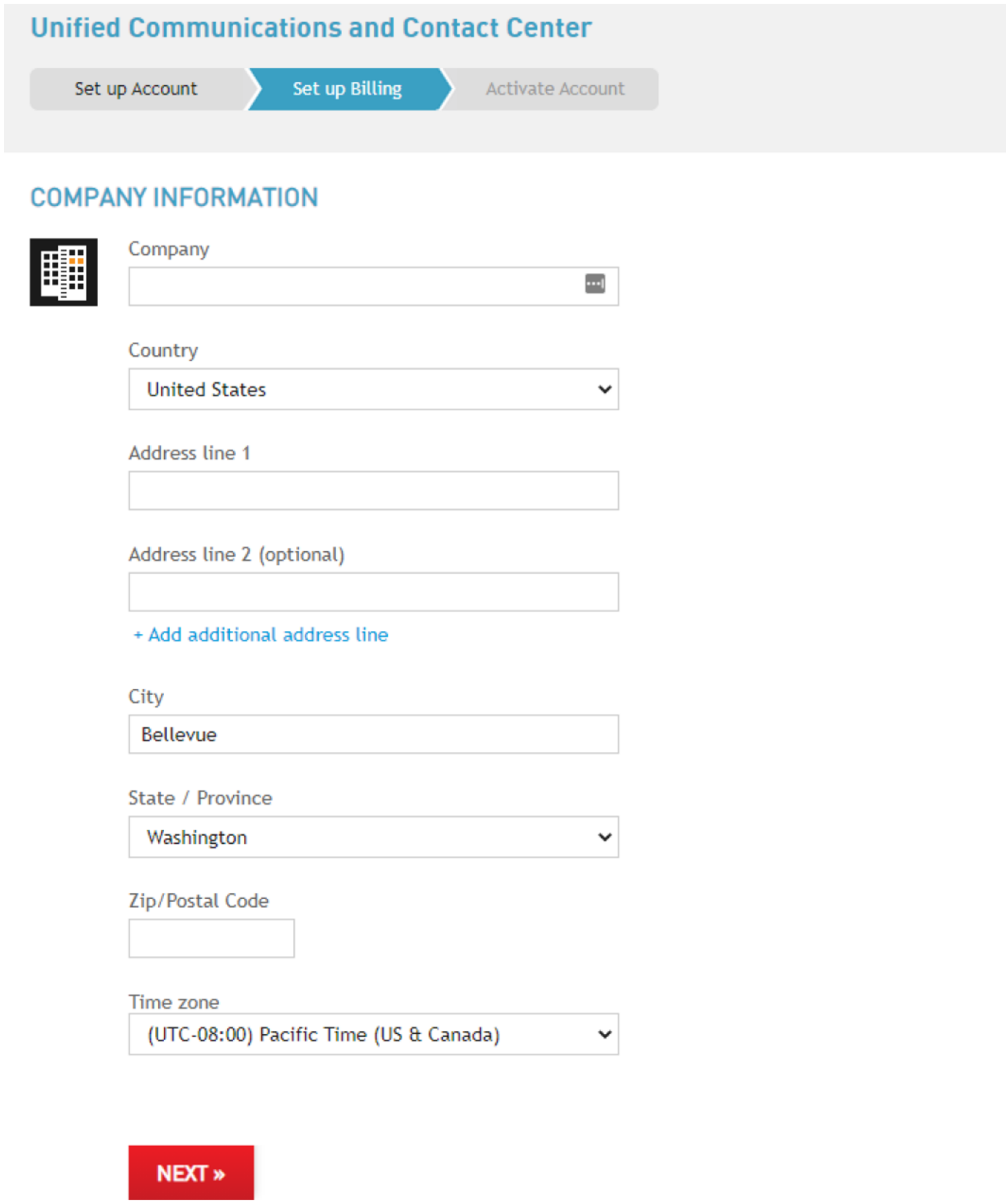

Figure 8 Company Information

3. Lastly, click the **Activate Account** button.

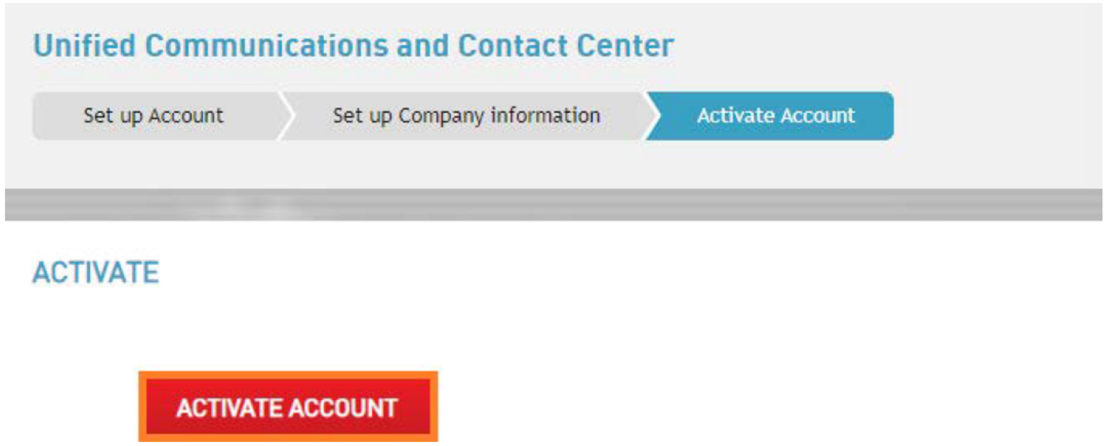

Figure 9 Activate Account

4. The account will be created in the Control Panel. The page will refresh once complete. To access the account in the Control Panel, click the **Get Started** button.

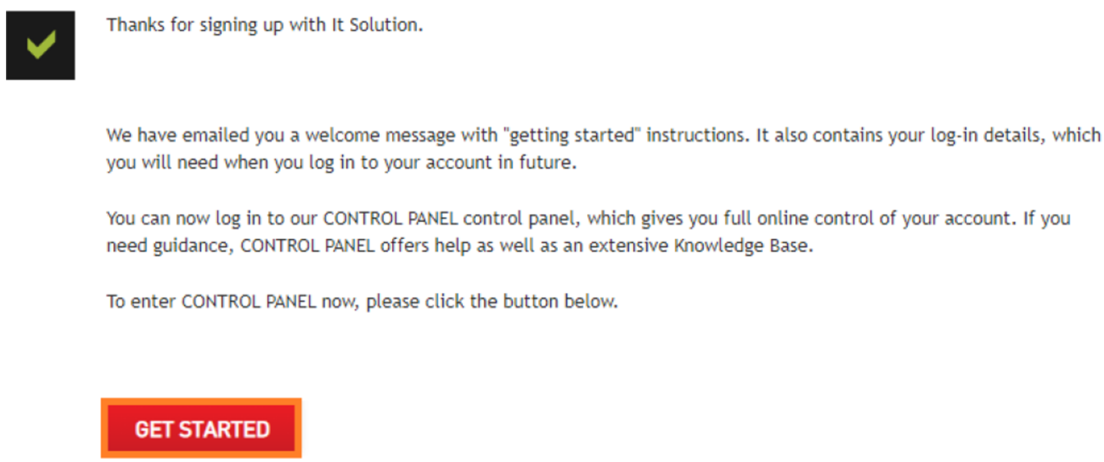

Figure 10 Get Started

1.4.4 Setting up the Account in the Control Panel

After creation, you may proceed to the next major phase of the overall CONNECT BRIDGE setup; which is the setup of the UNIVERGE BLUE account in the Control Panel.

1. The first step of setting up the account in the Control Panel is to install the CONNECT BRIDGE service. Refer to the information you gathered in the Deployment Planning section of this document and set the **Extension length** on this page to the same extension length as the on-premise PBX. Fill out the remainder of the page and select the **Get Started** button when complete.

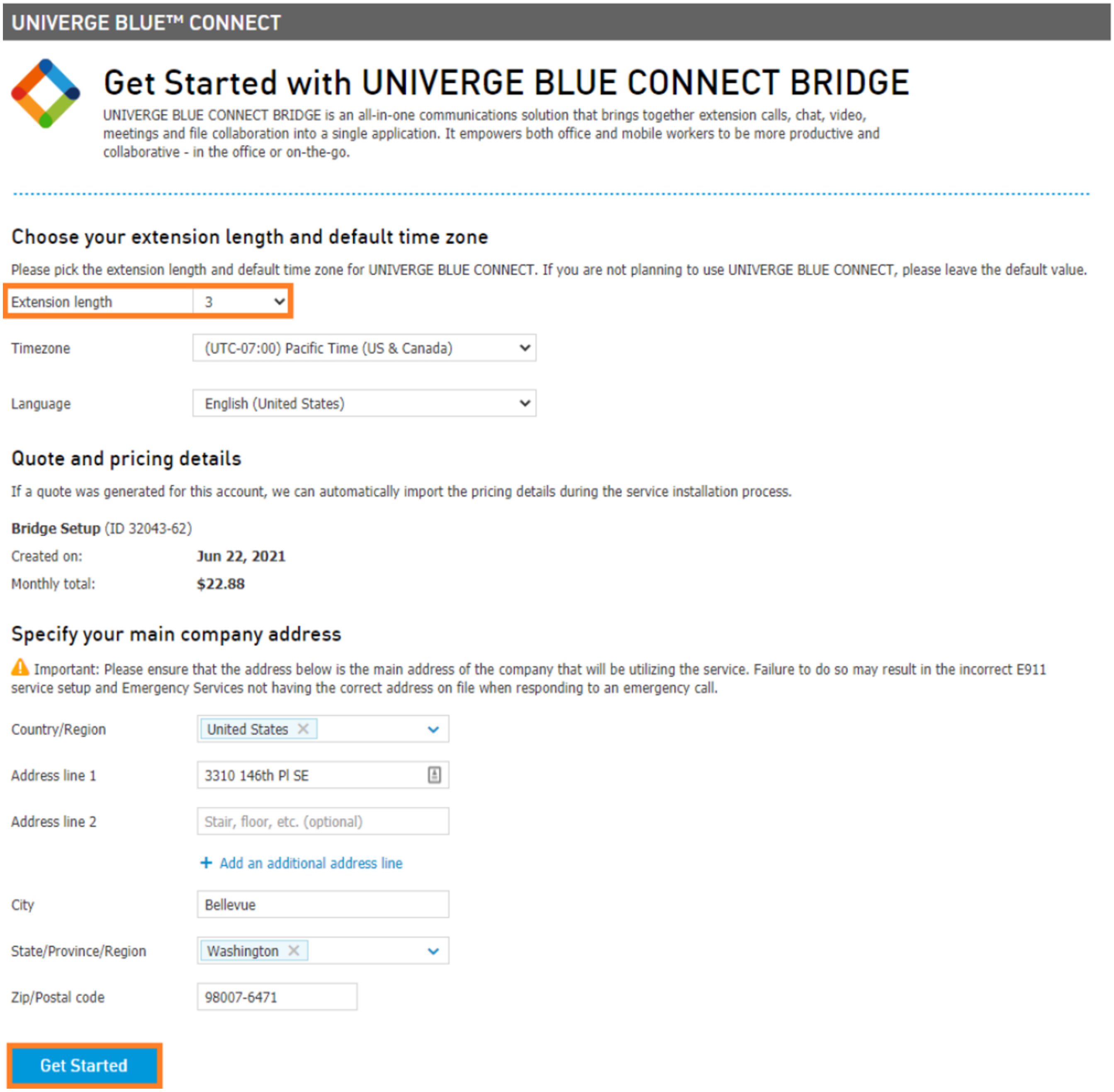

Figure 11 Extension Length

2. The Provisioning Wizard will launch automatically after installing the CONNECT BRIDGE service on the account.

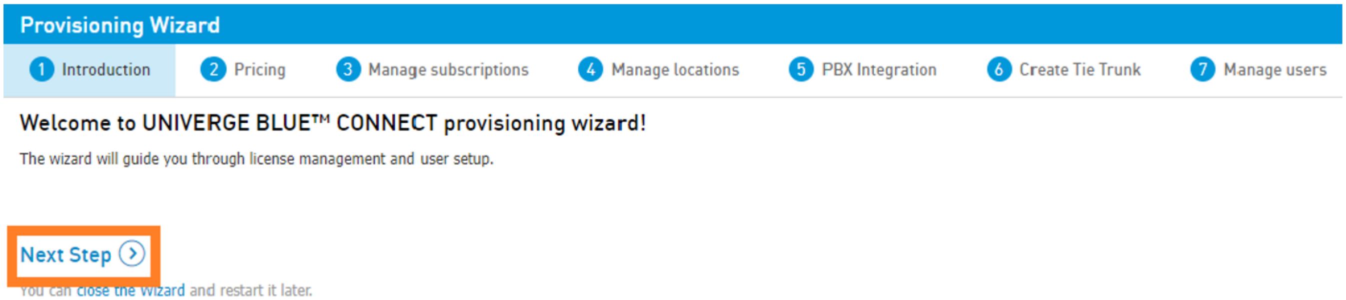

Figure 12 Provisioning Wizard – Introduction

3. Click **Next Step** and go to the Pricing page and set Customer Prices, if necessary.

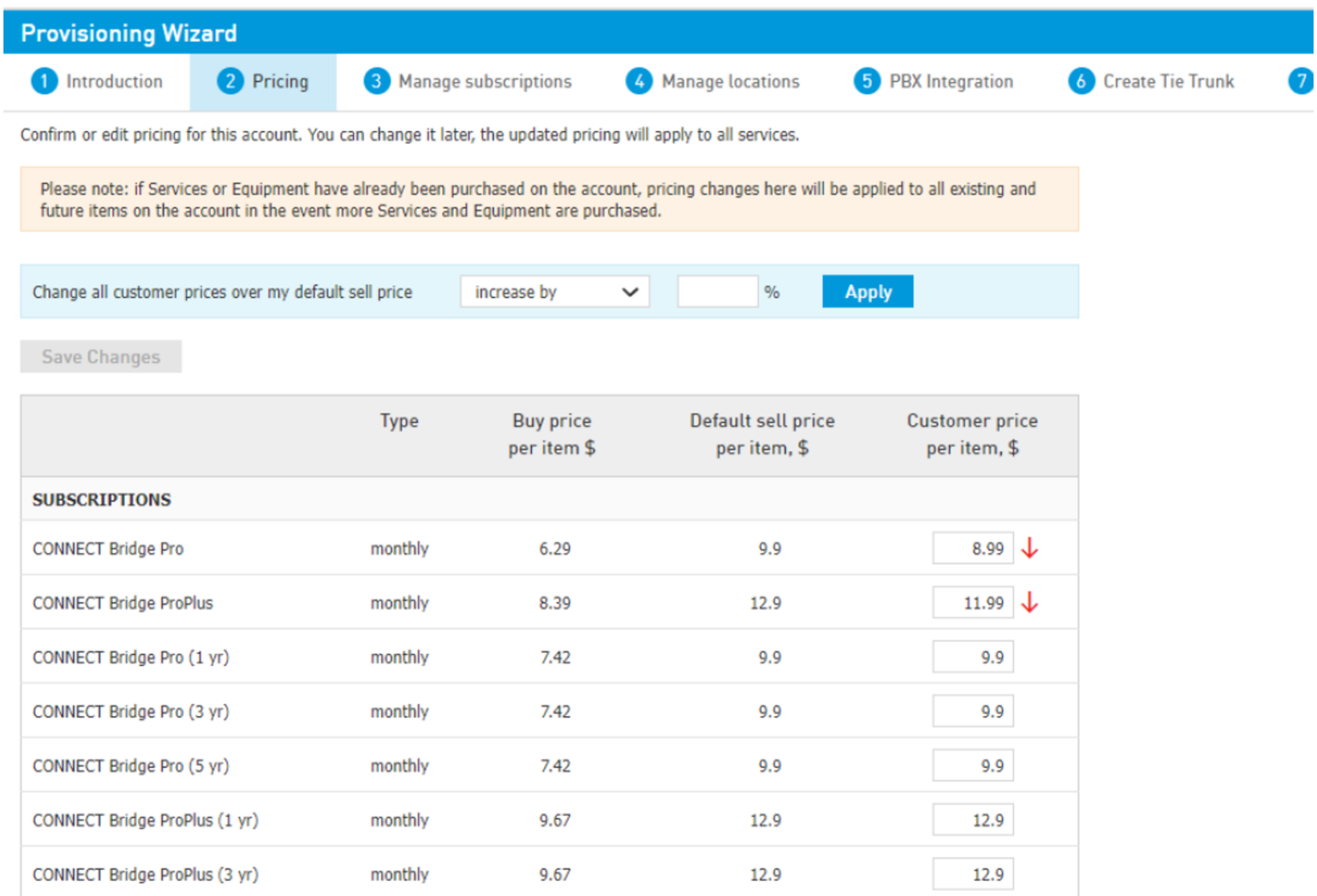

Figure 13 Provisioning Wizard – Pricing

4. Add at least 1 CONNECT BRIDGE or CONNECT BRIDGE ProPlus subscription to the account and click on **SAVE changes**. To proceed to the next step, click on **Next Step**.

   ✎ *NEC-J partners will only have one license called **CONNECT**. It is equivalent to CONNECT BRIDGE.*

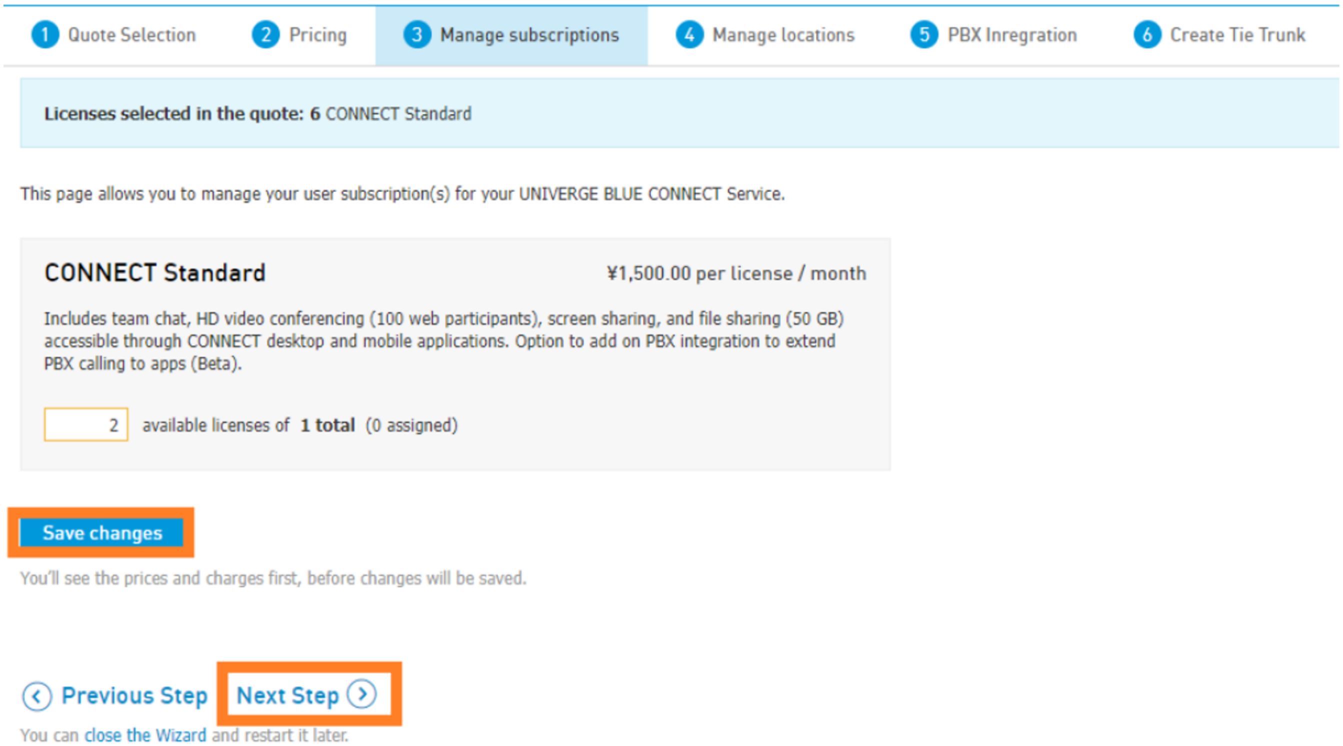

Figure 14 Provisioning Wizard – Manage Subscriptions

5. Add or update locations and location addresses on the **Manage locations** tab of the wizard then proceed to the **Manage users** tab.

6. On the PBX Integration tab, please read carefully and take note of the available resources in the Knowledge Base. Locate the PBX setup instructions for your PBX. You will need these instructions later in this setup when it is necessary to configure the on-prem PBX.

7. On the Create Tie Trunk tab, click **Create SIP Tie Trunk**. Refer to the information you gathered in the Deployment Planning section of this document and set the **On premise PBX type**, **Trunk access code**, **Concurrent call paths**, **Hardware key code** and **Codec priority** for this Tie Trunk and click **Save changes**. Take note of the **SIP Tie Trunk name** you use in this step. You will need it in one of the next steps.

   ✎ *NEC-J partners will not see the **Hardware licenses** section.*

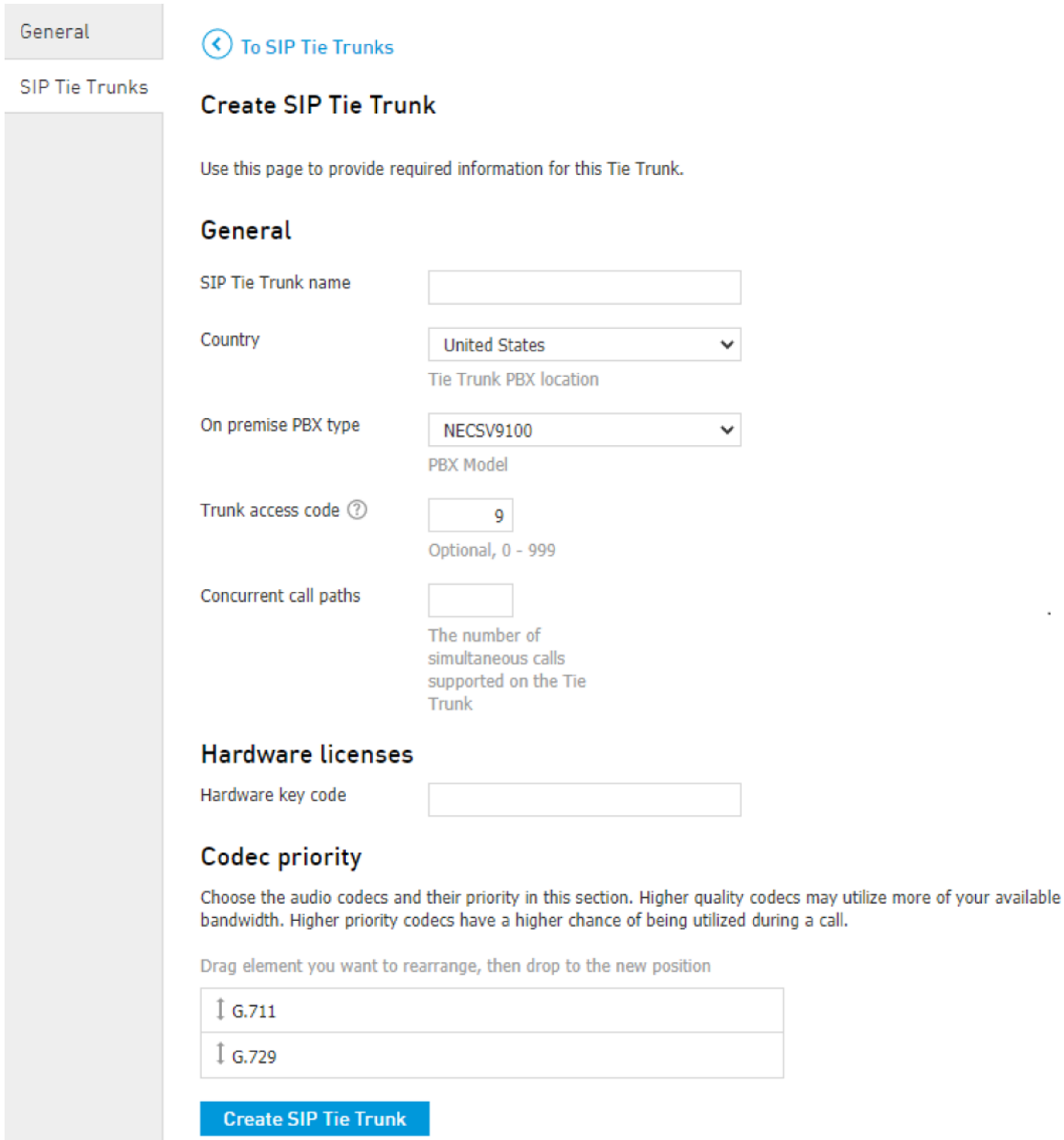

Figure 15 Create SIP Tie Trunk

8. From the **Manage users** tab, begin creating your Users. It is highly recommended to utilize the Mass User Import (MUI) feature to create an importable .csv file that contains all relevant User information. To use the MUI, select the link to **import multiple users**.

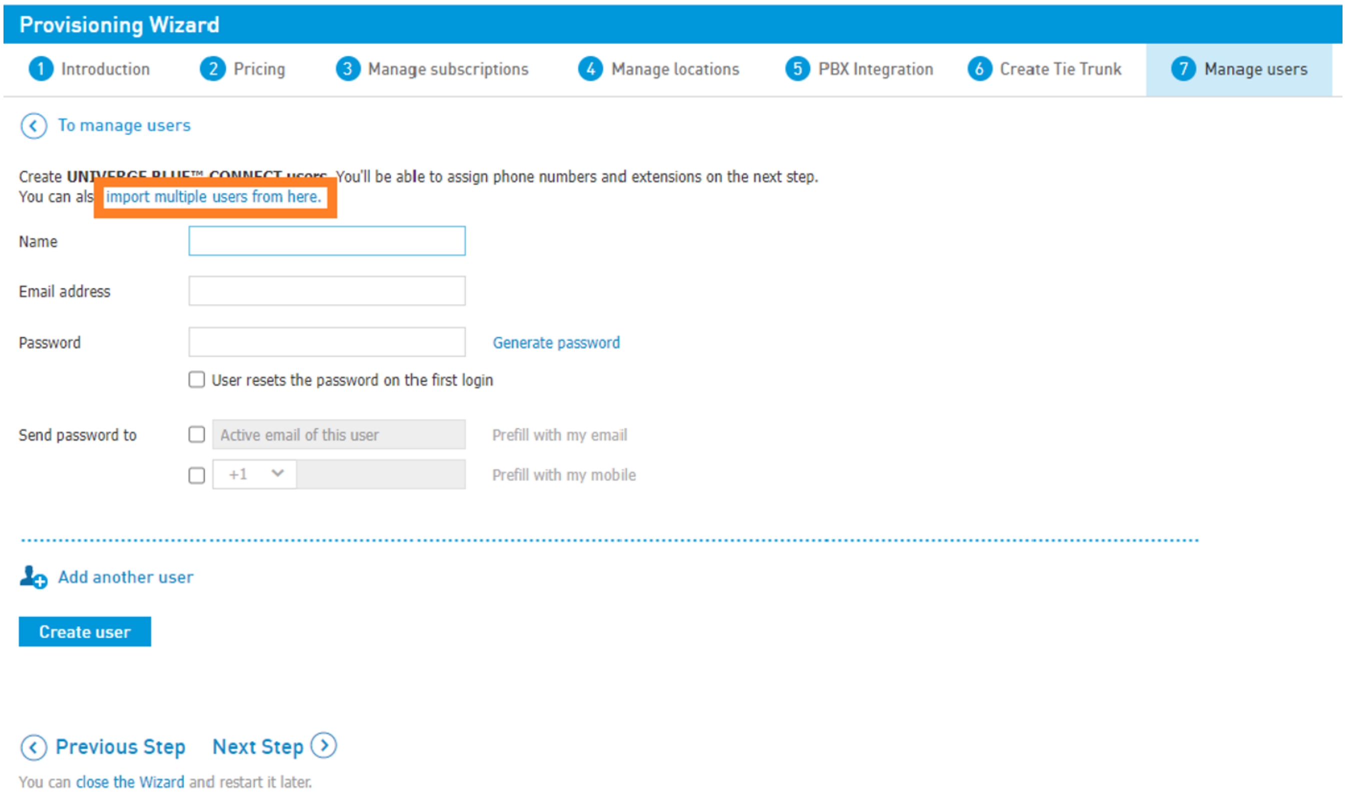

Figure 16 Provisioning Wizard – Manage Users

9. Click on the link to **Generate and download template** to obtain the .csv template file. When filling out the .csv file, refer to the information you gathered in the Deployment Planning section of this document. To fully setup your Bridge account, complete at least the following fields for each User in the .csv file:

Template Column	Example	Notes
Display Name	John Smith	
Email Address	jsmith@domain.com	
Subscription License Type	Connect Bridge Pro	From previous setup wizard step
Extension	1001	Use the User's Extension
Voicemail PIN	957643	
Cloud Extension	5001	
SIP Tie Trunk Name	SIP Trunk 1	From previous setup wizard step

10. Once filled out, click the link to **+ Add file** to upload the completed .csv file. Select where to send the passwords created for each User and click **Start import**. Note: in this step, many important things will occur, including:
    - Users will be created with the correct User extension
    - Users will be assigned a subscription
    - Users will have their Voicemail PIN set
    - Users will be assigned to the newly created Tie Trunk
    - Users will have the correct Cloud Extension assigned to them on the Tie Trunk

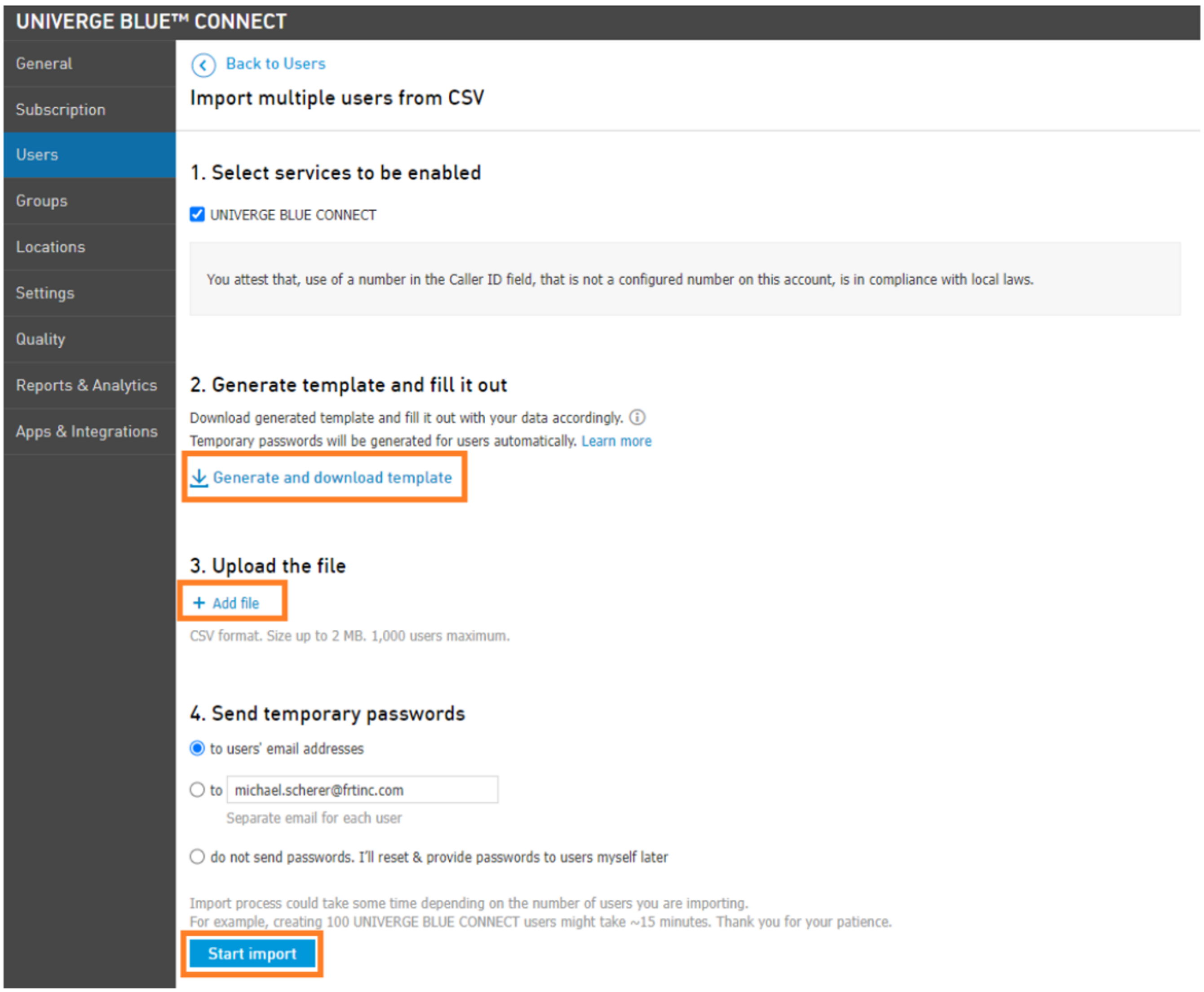

Figure 17 Import Multiple Users from CSV

11. On the last step, you will find the guides for deploying apps to your users.

12. When done, click the **Complete** button.

Congratulations, your UNIVERGE BLUE CONNECT BRIDGE account is well on its way to being setup.

1.4.5 Configuring the Tie Trunk

A Tie Trunk was created by using the Provisioning Wizard, but the Tie Trunk may still require further configuration, such as adding Premise Extensions. To begin the process of creating a Tie Trunk, navigate to **Settings** > **SIP Tie Trunks**.

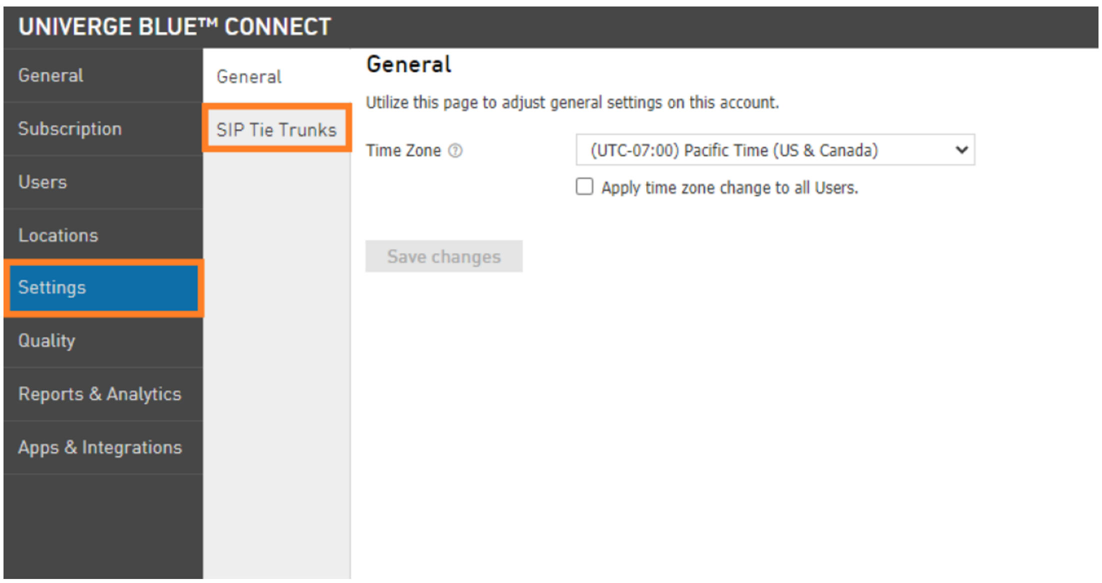

Figure 18 Settings – SIP Tie Trunks

Locate the Tie Trunk you wish to edit and click on its name.

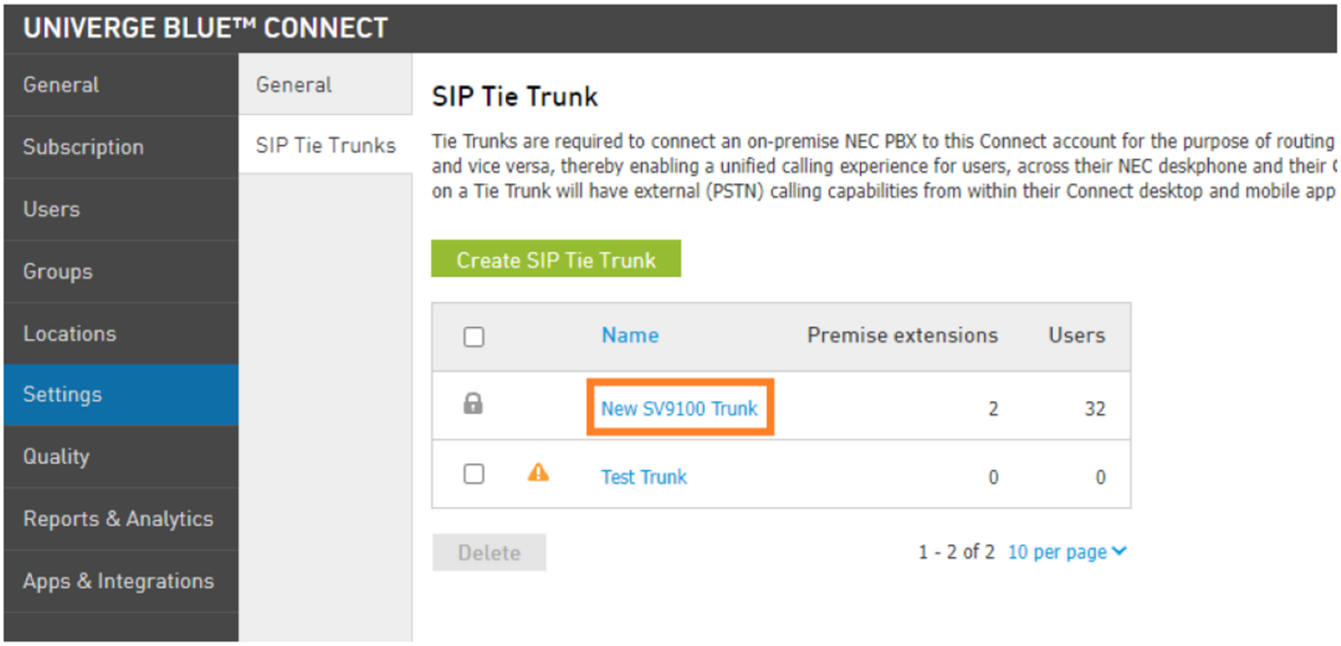

Figure 19 SIP Tie Trunks – Create SIP Tie Trunk

1.4.6 Adding Premise Extensions to the Tie Trunk

Any valid Premise Extension that is in use needs to be added to the Tie Trunk, so the UNIVERGE BLUE account understands to route calls to these extensions over the Tie Trunk to the on-premise PBX. Please refer to the information you gathered in the Deployment Planning section of this document in this section.

1. Navigate to the **Premise Extensions** tab of the Tie Trunk and click on **Add premise extensions**.

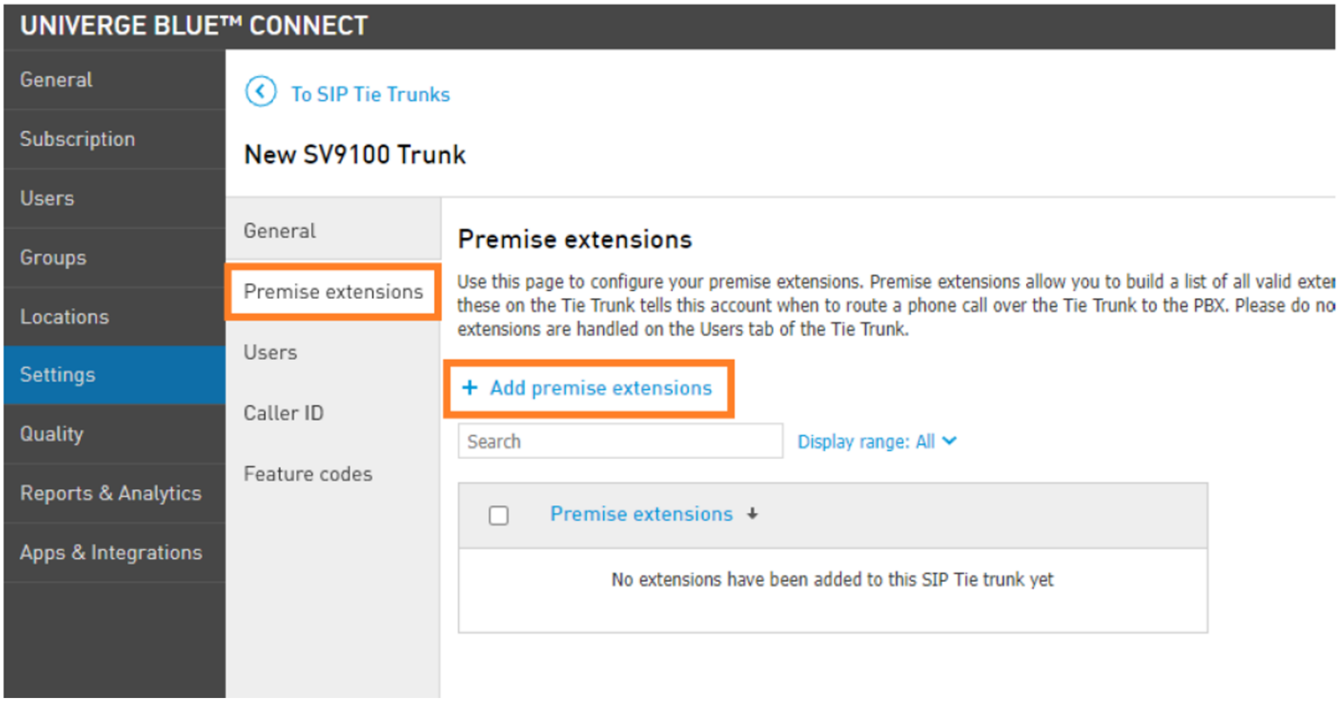

Figure 20 Settings – Premise Extensions

2. You may select to add single extensions, one at a time, or you may select to add ranges of extensions. Choose the option that works best for your situation.

3. To add extensions one at a time, select the option for **Single extension** and begin typing your single Premise Extensions into the field provided. When finished, click **Add premise extensions**.

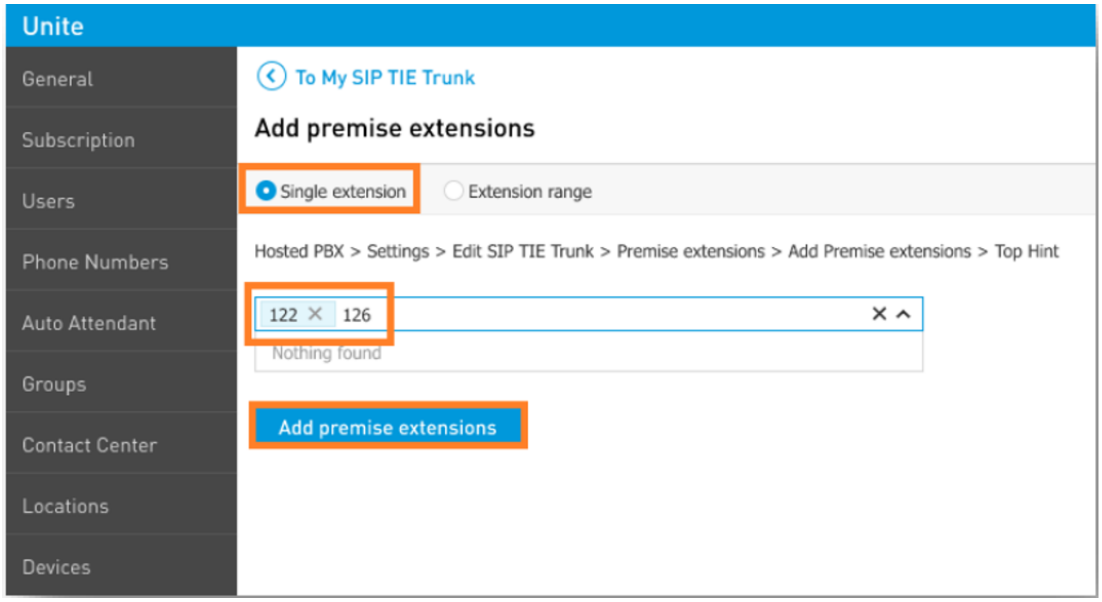

Figure 21 Add Premise Extensions – Single Extension

4. To add extensions in a range, select the option for **Extension range** and enter the start and end extensions in the range, and click **Add trunk extensions**.

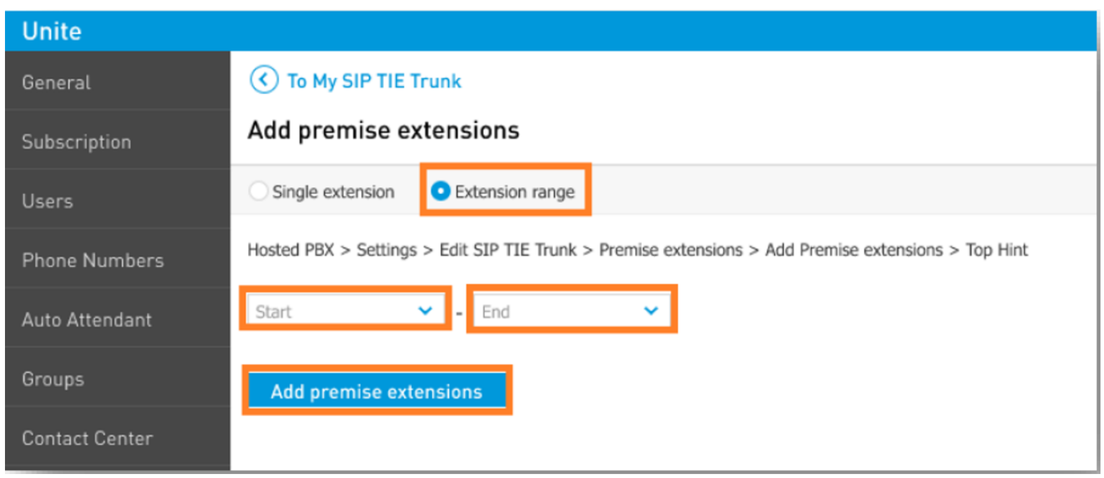

Figure 22 Add Premise Extensions – Extension Range

5. Once fully added, premise extensions may be managed as necessary.

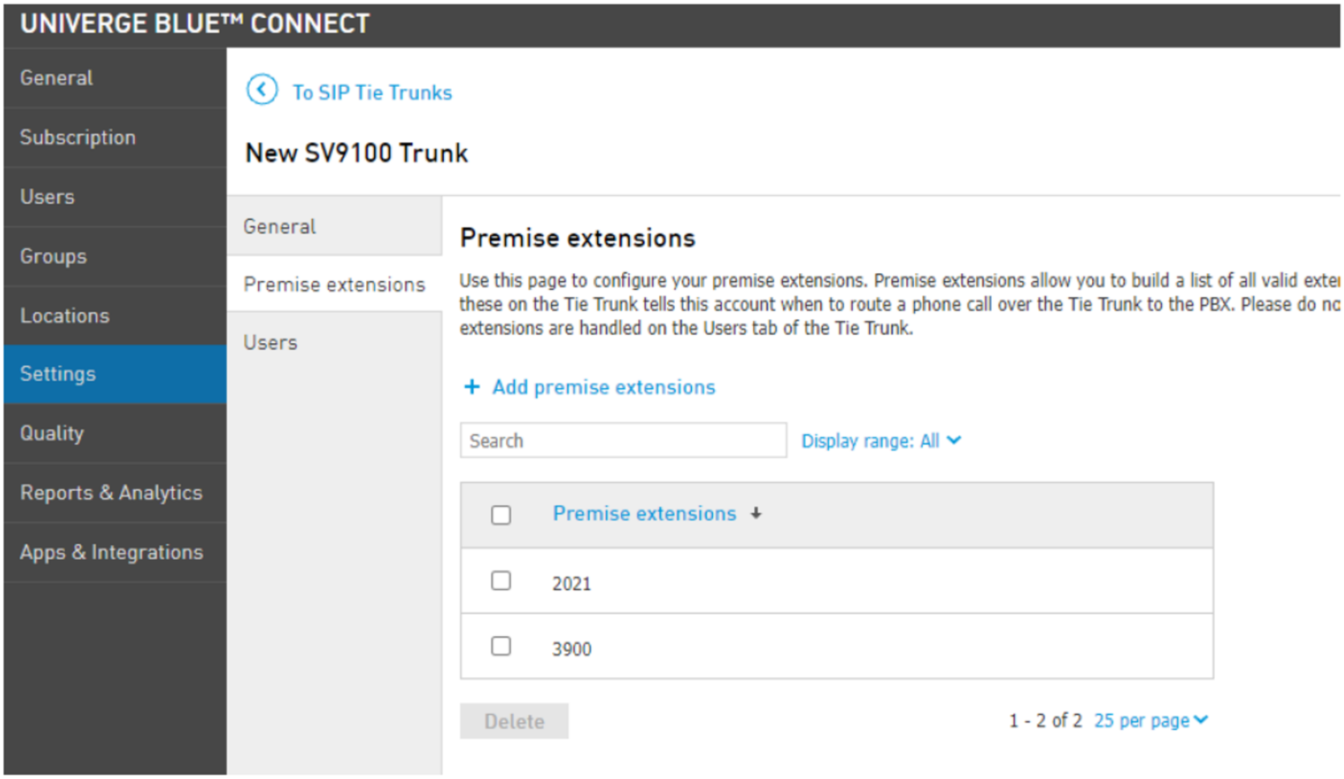

Figure 23 Premise Extensions

1.4.7 Configuring Caller ID on the Tie Trunk

By default, calls sent from the CONNECT BRIDGE account to the on-prem PBX over the Tie Trunk always send the User's Extension as the caller ID. If it is desired to better differentiate within your call history and/or CDR records where the call is coming from, you may toggle this setting to use the user's Cloud Extension as their caller ID instead.

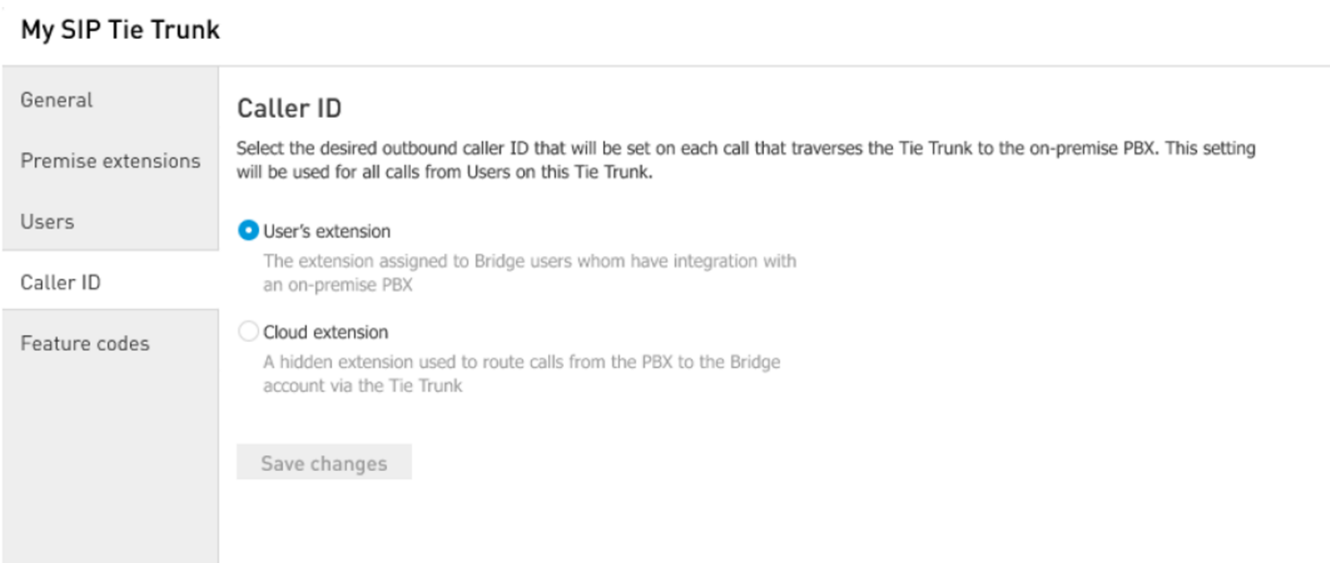

Figure 24 My SIP Tie Trunk – Caller ID

1.4.8 Configuring Feature Codes

The Feature Codes feature of the Tie Trunk allows you to program specific dial strings that will be sent across the Tie Trunk to the on-prem PBX when dialed from the CONNECT BRIDGE apps. Feature Codes work similarly to Premise Extensions, in that this feature allows you to program specific situations when a call will be sent across the Tie Trunk to the on-prem PBX. To utilize this feature, simply dial the programmed dial strings like you would a telephone number from the CONNECT BRIDGE apps.

To begin, click on the link to **Add dial string**.

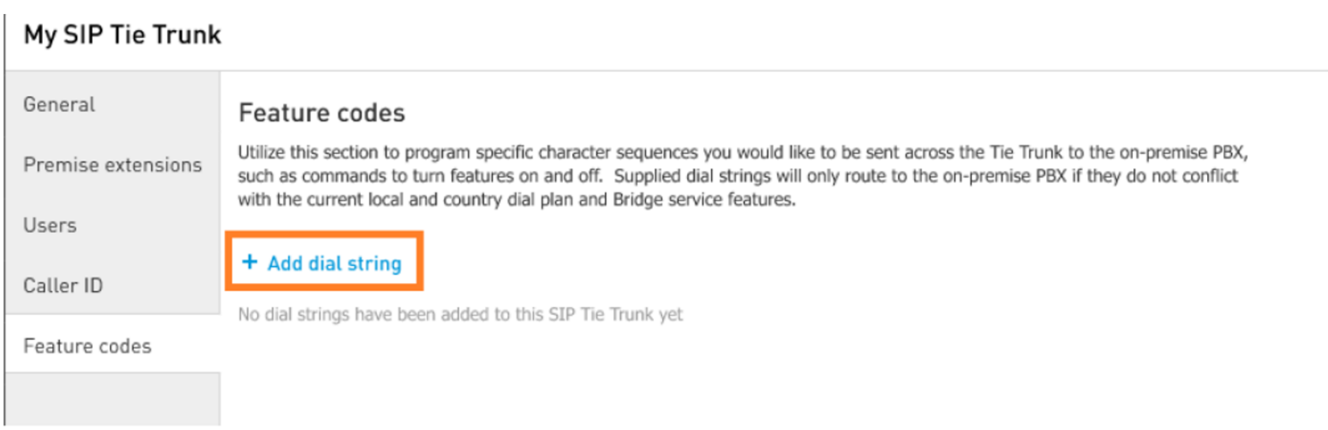

Figure 25 My SIP Tie Trunk – Feature Codes

Dial strings may be either:

- Full match – this option requires a complete match of the dial string before it will send to the on-prem PBX. Ex. Program ***551234**. *551234 would have to dialed in full before the dial string will be sent to the on-prem PBX.
- Starts with – this option requires the dial string to begin with all programmed digits before it will send to the on-prem PBX. Ex. Program ***55**. *55 followed by any other digits will be sent to the on-prem PBX.

Select the Type of dial string you wish to create, then enter the actual dial string required. Add an additional comment if necessary and click **Add dial string**.

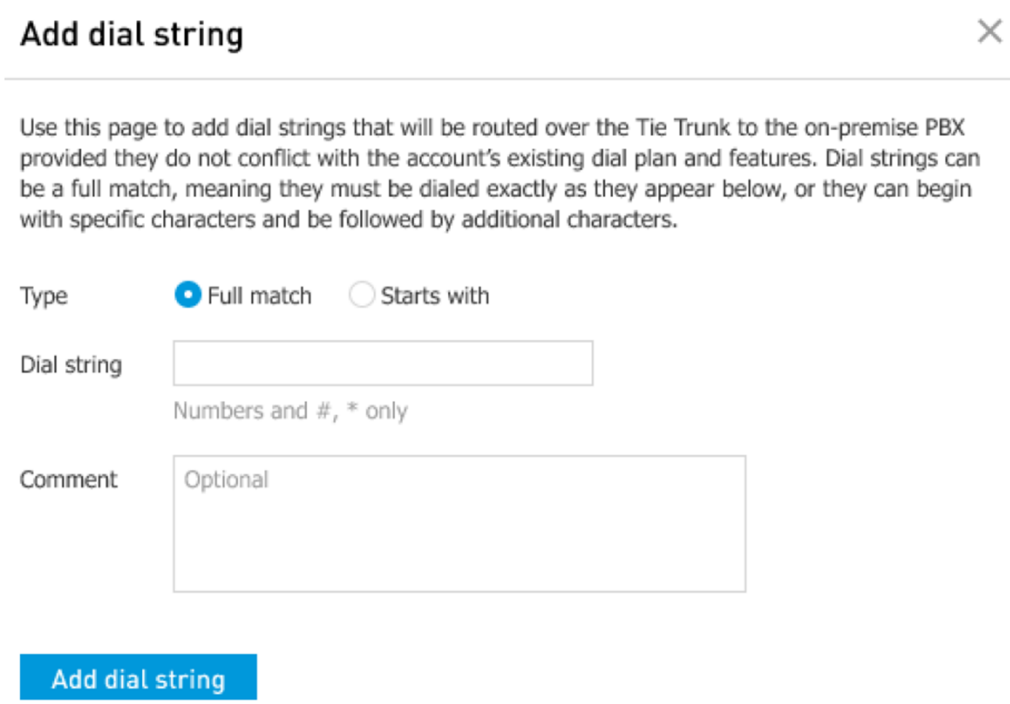

Figure 26 Add Dial String

1.4.9 Networking and Firewall Information for UNIVERGE BLUE Cloud

Various parts of CONNECT BRIDGE need to communicate through the firewall at each location where service is installed. Please refer to the following knowledge base article for information on how to configure a firewall for CONNECT BRIDGE: https://kb.univerge.blue/en-US/Article/38504

SECTION 2 SV9300 SETUP FOR CONNECT BRIDGE

2.1 Requirements

SV9300 system software v8.3.0 or higher.

However, use of SV9300 system software v9.2.1 or higher is recommended as this removes looped connection on a Tie Trunk. See Appendix 1 Section 2 Limitations for more details.

With the SV9300, a is required in addition to licensing for IP (SIP) trunks.

A minimum of four IP (SIP) trunks are required due to the NEC Communications Server infrastructure setup.

- SV9300 CPU software version V8.3.0 or higher
- VoIP gateway daughter board
- SIP Trunk Port Licenses (min 4)
- VOIPDB (PZ-64IPLC-A, PZ-128IPLC-A, GPZ-64IPLD or GPZ-128IPLD)
- SV93 MOBILITY ACCESS-1 LIC (1 for each cloud extension)

2.2 Limitations

The following limitations apply:

- UNIVERGE SV9300 supports T.38 or G.711 pass-through faxing.
  - Supported fax protocol is determined by SIP trunk provider. If SIP trunk provider only supports best-effort, SIP trunk faxing may be unreliable.
- SIP diversion header is not supported – Call forwarding to 8xx numbers
- SIP Privacy – Cannot mark the calling party number as private or restricted
- Secondary SIP server for failover is not supported

2.3 SV9300 PBX Programming

2.3.1 System Version and License Check

Values shown are for example purposes only. Your actual IP values will be determined by your local LAN administrator.

After connecting, check the SV9300 software version.

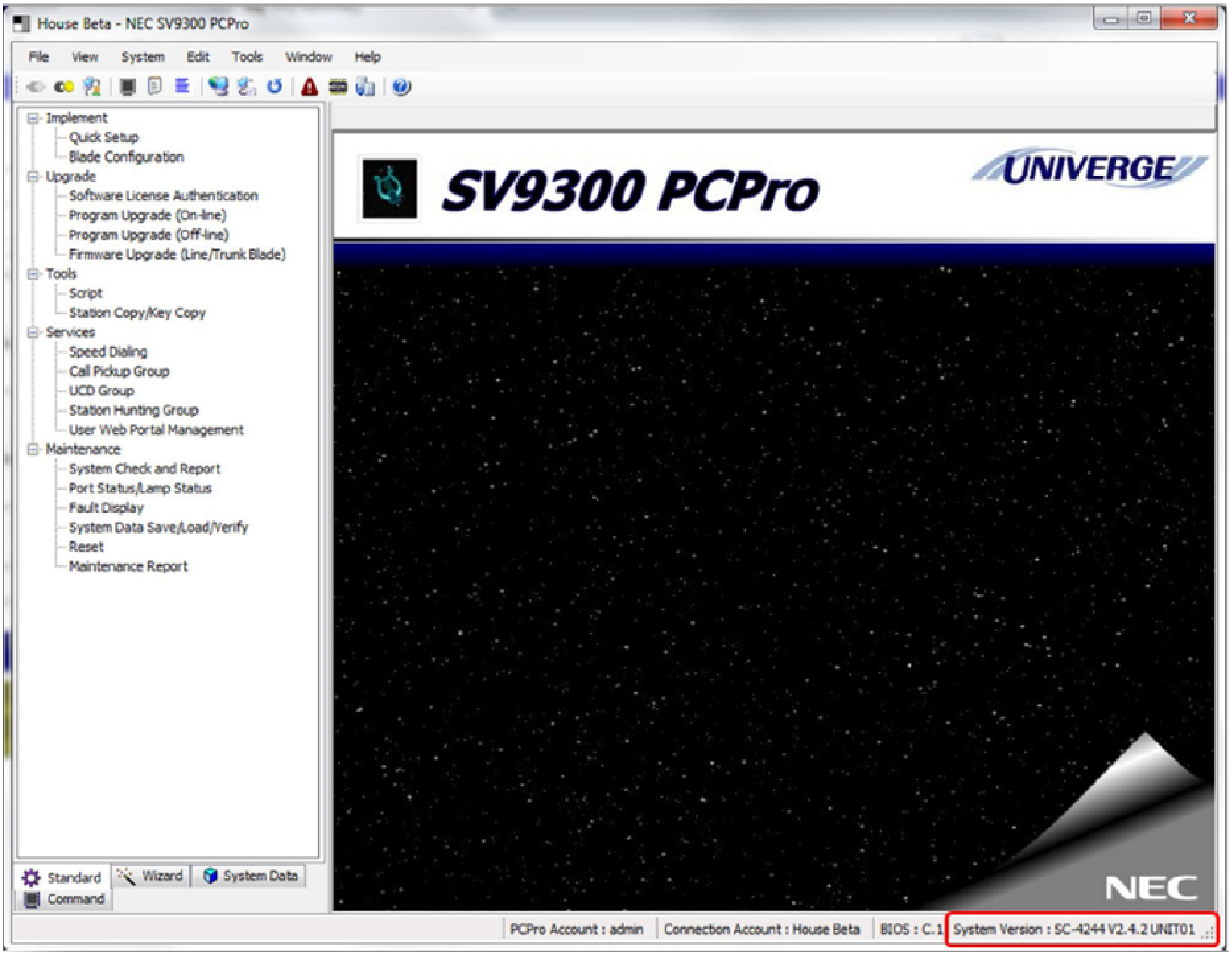

Figure 27 System Version

Check the SIP Trunk Port Licenses

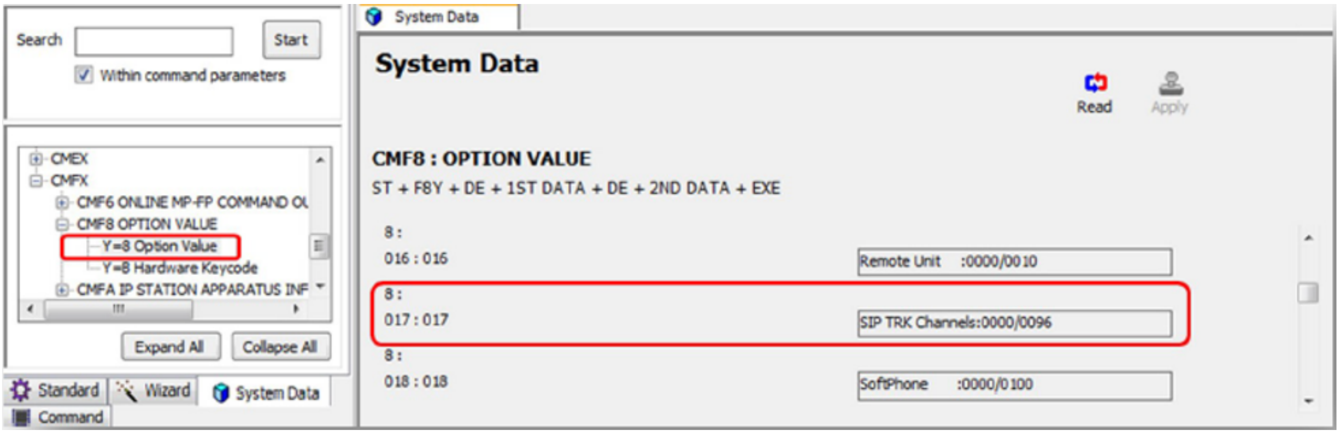

Figure 28 CM F8 Option Value

2.4 CC-CP10A Network Setup

Values shown are for example purposes only. Your actual IP values will be determined by your local LAN administrator.

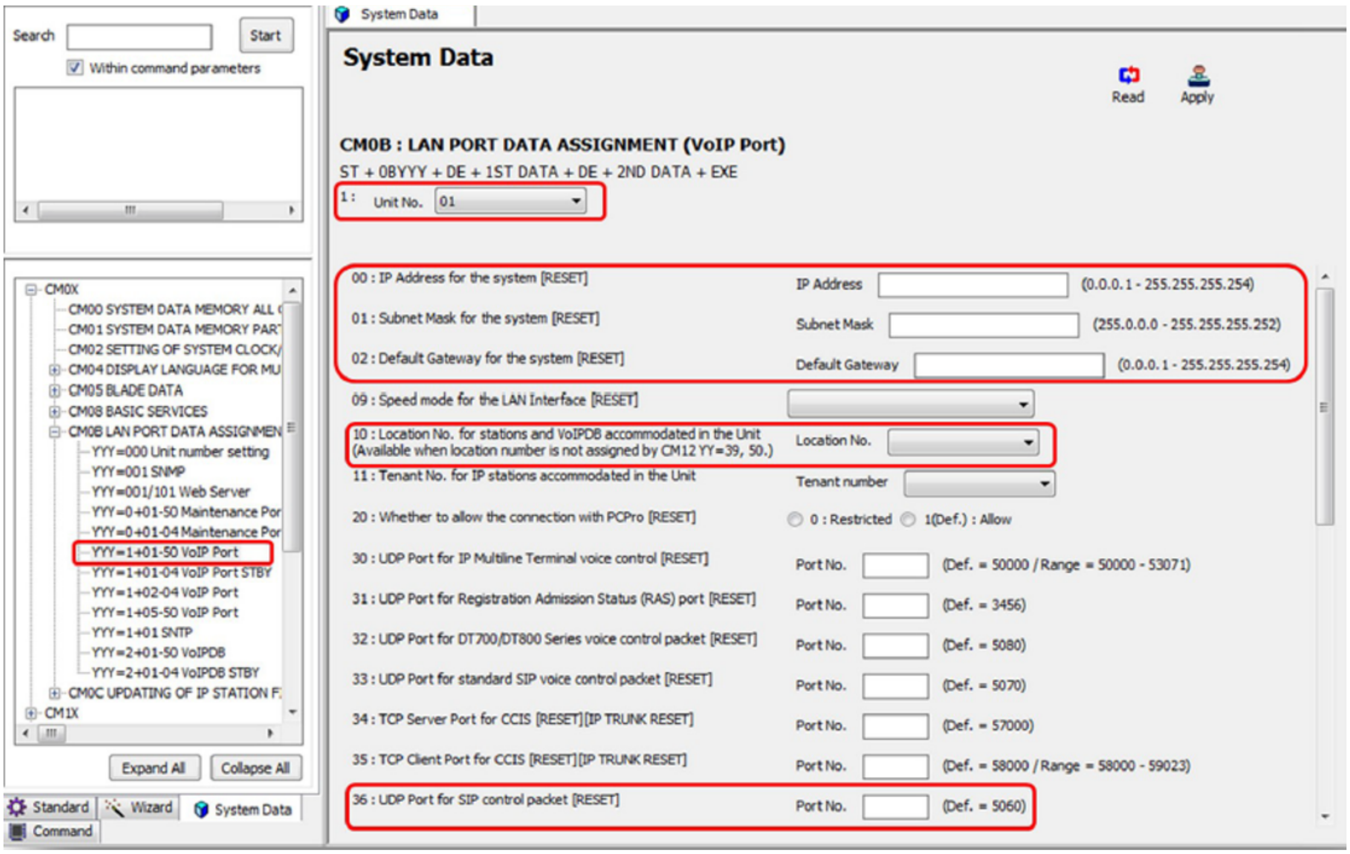

Figure 29 CM 0B Network Setup

1. Select SV9300 Main/Remote Unit No. accommodating SIP Trunk Channels, then click **Read** to get the current data settings.
2. FD=00 Enter SV9300 VoIPDB IP Address.
3. FD=01 Enter SV9300 VoIPDB Subnet Mask.
4. FD=02 Enter SV9300 VoIPDB Default Gateway Address.
5. FD=10 Set VoIPDB location number.

   Location number is used for SIP trunk T.38 fax service. NEC is unable to support best effort faxing do to unreliable fax communications.
6. FD=36 Leave Blank to use default port 5060 for SIP trunk control packets.
7. Click **Apply**.

Command Line Example: Unit 01

CM 0B101>00>172.24.142.55	(Unit 01 VoIP Address)
CM 0B101>01>255.255.255.0	(Unit 01 VoIP Subnet)
CM 0B101>02>172.24.142.1	(Unit 01 VoIP Default GW)
CM 0B101>10>NONE	(Default: Location 00)
CM 0B101>36>	(Default: SIP Trunk Port 5060)

The SV9300 must be reset in order for the change to take effect.

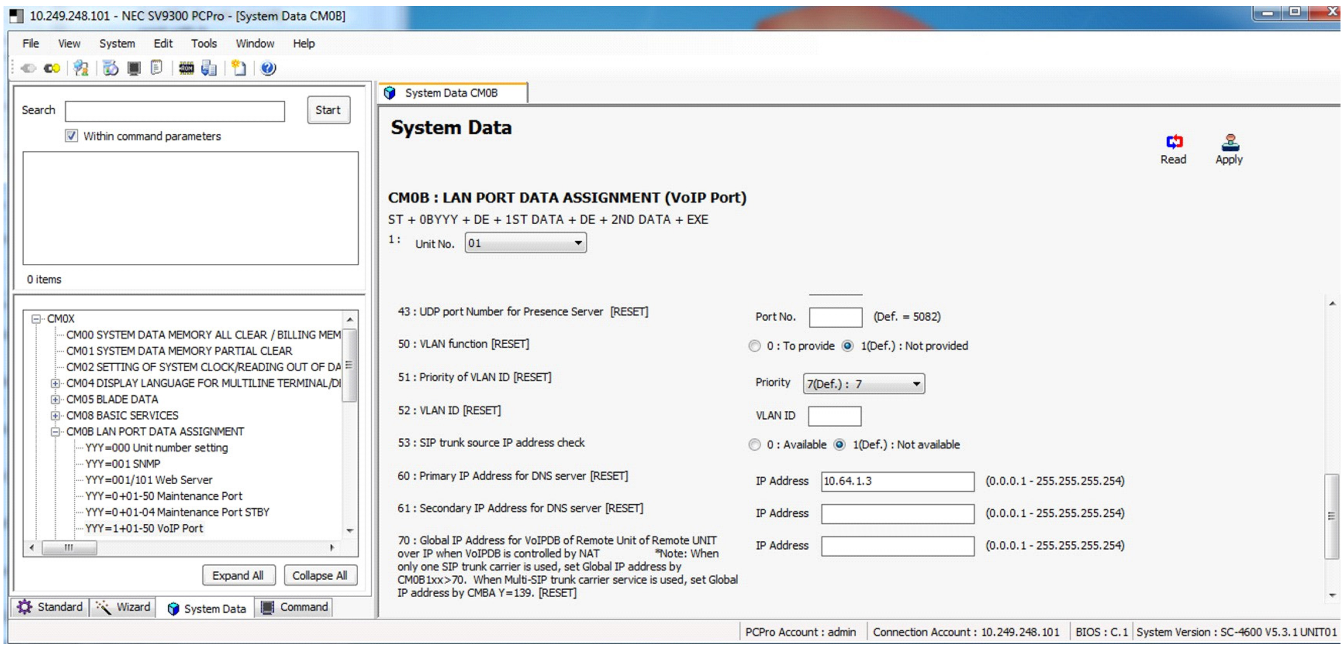

Figure 30 CM 0B Network (Continued)

1. Select SV9300 Main/Remote Unit No. accommodating SIP Trunk Channels, then click **Read** to get the current data settings.
2. FD=60 Enter primary DNS server IP Address (optional).

   FD=61 Enter secondary DNS server IP Address (optional).

   FD=70 Enter global IP Address (optional).

   If more than 1 profile is used enter global IP address in CM BA139.
3. Click **Apply**.

Command Line Example: Unit 01

CM 0B101>60>151.164.1.8	(Unit 01 Primary DNS Server Address)
CM 0B101>61>151.164.11.201	(Unit 01 Secondary DNS Server Address)
CM 0B101>70>172.24.142.1	(Unit 01 Global IP Address)

The SV9300 must be reset in order for the change to take effect.

2.5 IP PAD Network Settings

Values shown are for example purposes only. Your actual IP values will be determined by your local LAN administrator.

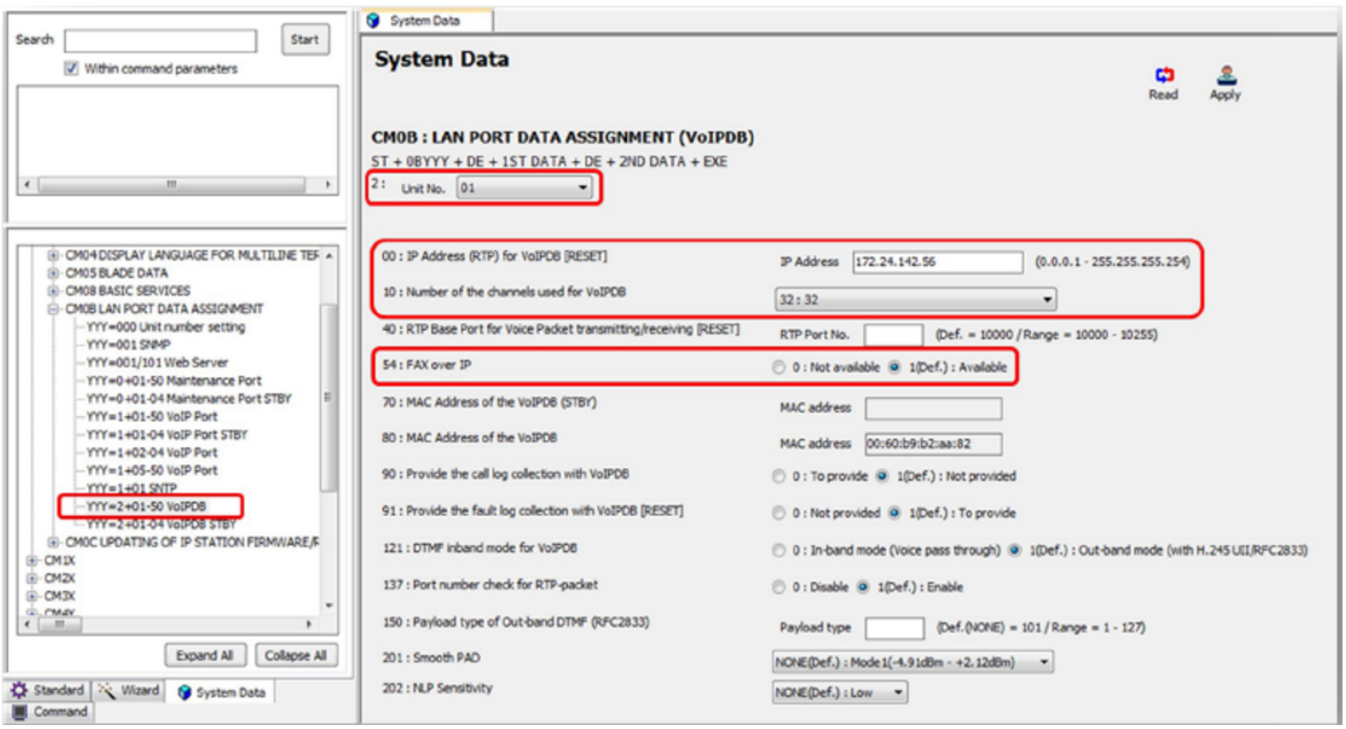

Figure 31 CM 0B IP PAD Network Settings

1. Select SV9300 Main/Remote Unit No. accommodating SIP Trunk Channels, then click **Read** to get the current data settings.
2. FD=00 Enter VoIP IPPAD IP address.

   FD=10 Select number of VoIP IPPAD channels.
3. FD=54 Select 1(Def.) : Available (to allow Fax over IP).
4. Click **Apply**.

Command Line Example: Unit 01

CM 0B201>00>172.24.142.56	(Unit 01 VoIP IPPAD Address)
CM 0B201>10>32	(Unit 01 VoIP IPPAD Channels)
CM CM 0B201>54>1	(Unit 01 Allow Fax over IP)

The SV9300 must be reset in order for the changes to take effect.

2.6 SIP Trunk Port Allocation

All values shown are for example purposes only. Your actual IP values will be determined by your implementation team.

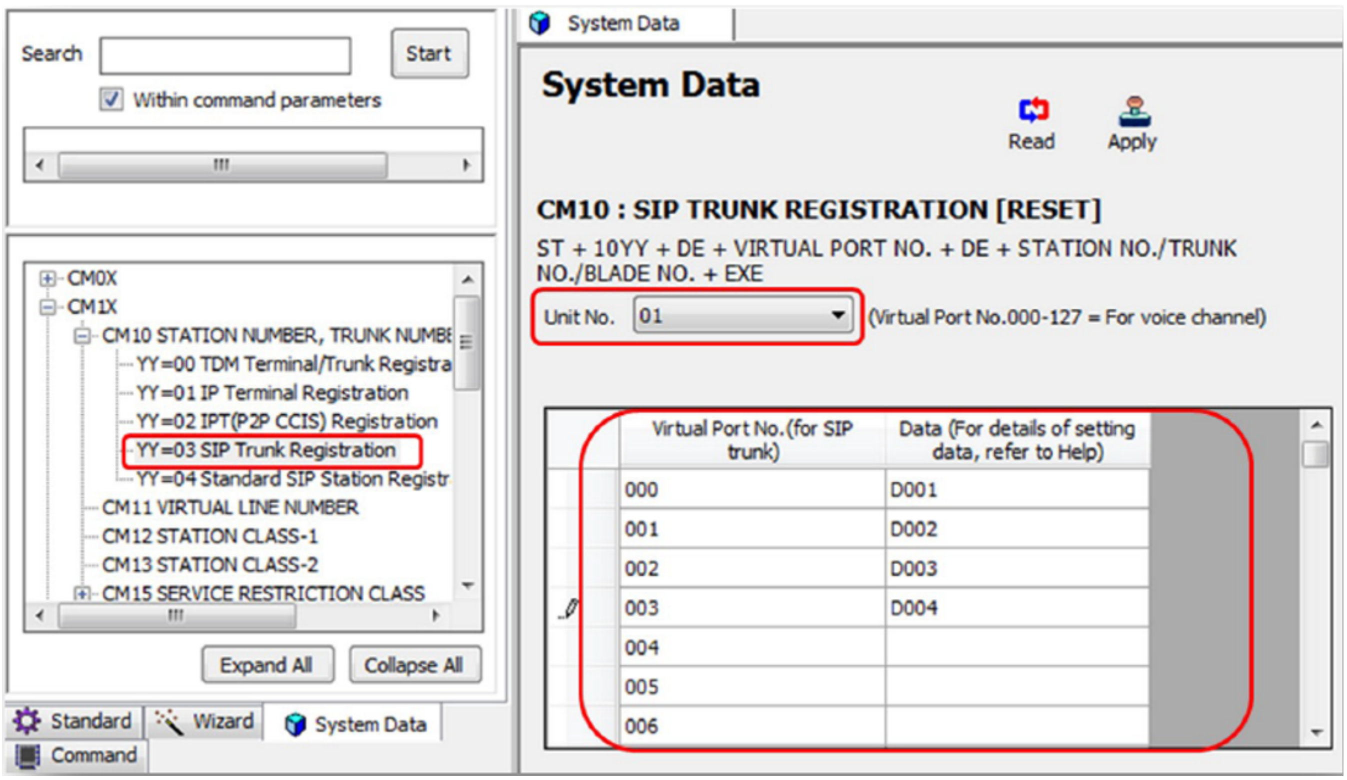

Figure 32 CM 10 SIP Trunk Port Allocation

1. Select SV9300 Main/Remote Unit No. accommodating SIP Trunk Channels, then click **Read** to get the current data settings.
2. Enter Trunk Numbers used for the voice channels.
3. Click **Apply**.

Command Line Example: Unit 01

CM 1003>01000>D001	(Unit 01 SIP Trunk Channel 1)
CM 1003>01001>D002	(Unit 01 SIP Trunk Channel 2)
CM 1003>01002>D003	(Unit 01 SIP Trunk Channel 3)
CM 1003>01003>D004	(Unit 01 SIP Trunk Channel 4)

2.7 SIP Trunk Port Settings

All values shown are for example purposes only. Your actual IP values will be determined by your implementation team.

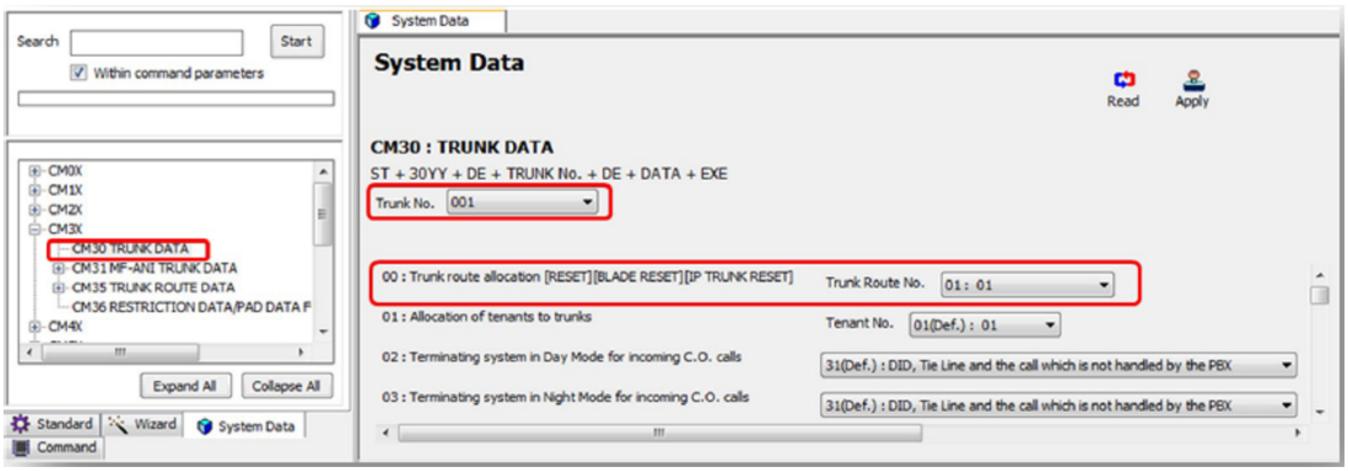

Figure 33 CM 30 SIP Trunk Port Settings

1. Select each Voice Channel and click the **Read** button to get the current data settings.
2. YY=00 Assign the same Trunk Route Number to each voice channel.

   If multiple SIP trunk servers are in service, separate routes must be used for each server.
3. Click **Apply**.

Command Line Example: Route 01

CM 3000>001>01	(Voice Trunk Route Number 01)
CM 3000>002>01	(Voice Trunk Route Number 01)
CM 3000>003>01	(Voice Trunk Route Number 01)
CM 3000>004>01	(Voice Trunk Route Number 01)

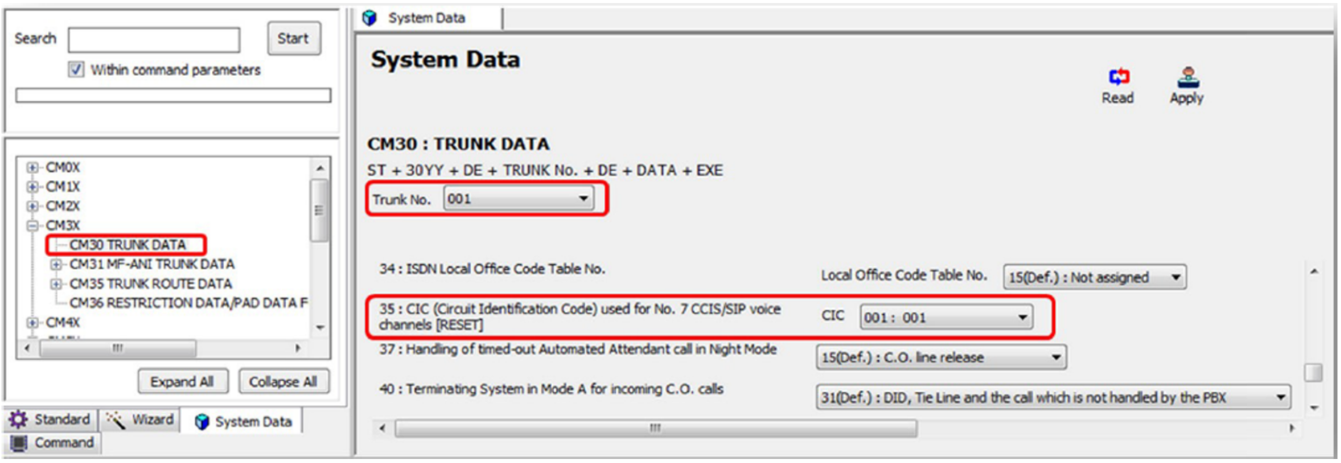

Figure 34 CM 30 SIP Trunk Port Settings (continued)

1. Select each Voice Channel and click the **Read** button to get the current data settings.
2. YY=35 Assign CIC number to each voice trunk.

   CICs should be assigned in order of voice trunks (i.e. trunk 001= CIC 001, trunk 002=CIC 002 …).

   For each SIP trunk server CICs will start at 001 (i.e. SIP trunk server 1 uses CICs 001, 002, … and SIP trunk server 2 will use CICs 001, 002, …).
3. Click **Apply**.

Command Line Example:

CM 3035>001>001	(Assign CIC to Channel 1)
CM 3035>002>002	(Assign CIC to Channel 2)
CM 3035>003>003	(Assign CIC to Channel 3)
CM 3035>004>004	(Assign CIC to Channel 4)

2.8 SIP Trunk Route Settings

All values shown are for example purposes only. Your actual IP values will be determined by your implementation team.

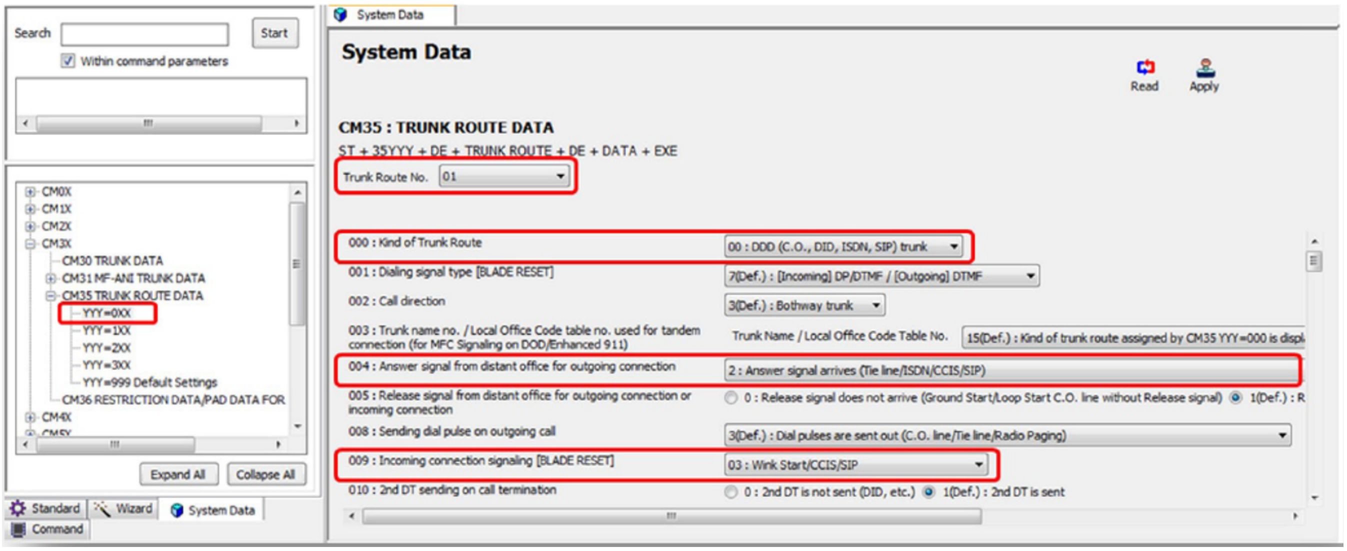

Figure 35 CM 35 SIP Trunk Route Settings

1. Select the Trunk Route Number assigned to voice channels and click the **Read** button to get current data settings.
2. YYY=000 Select 00 for SIP trunk service.
3. YYY=004 Select 2 for SIP trunk service.
4. YYY=009 Select 03 for SIP trunk service.
5. Click **Apply**.

Command Line Example:

CM 35000>01>00	(SIP Trunk Route Data: Provide SIP Service)
CM 35004>01>2	(SIP Trunk Route Data: Provide SIP Service)
CM 35009>01>03	(SIP Trunk Route Data: Provide SIP Service)

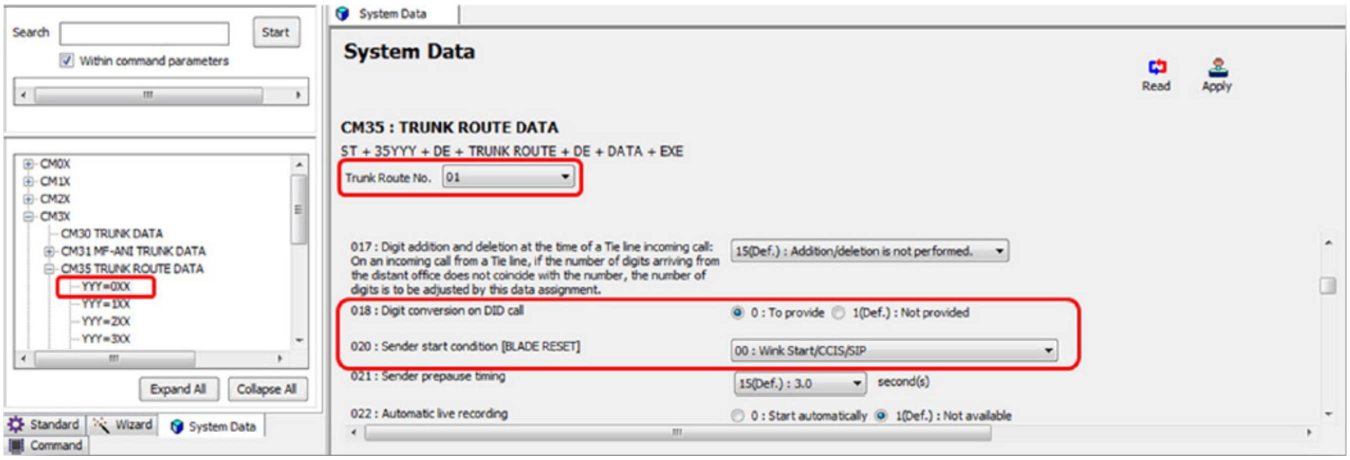

Figure 36 CM 35 SIP Trunk Route Settings (Continued)

1. Select the Trunk Route Number assigned to voice channels and click the **Read** button to get current data settings.
2. YYY=018 Select 0 for DID digit conversion.

   YYY=020 Select 00 for SIP trunk service.
3. Click **Apply**.

Command Line Example:

CM 35018>01>0	(SIP Trunk Route Data: Provide DID Digit Service)
CM 35020>01>00	(SIP Trunk Route Data: Provide SIP Service)

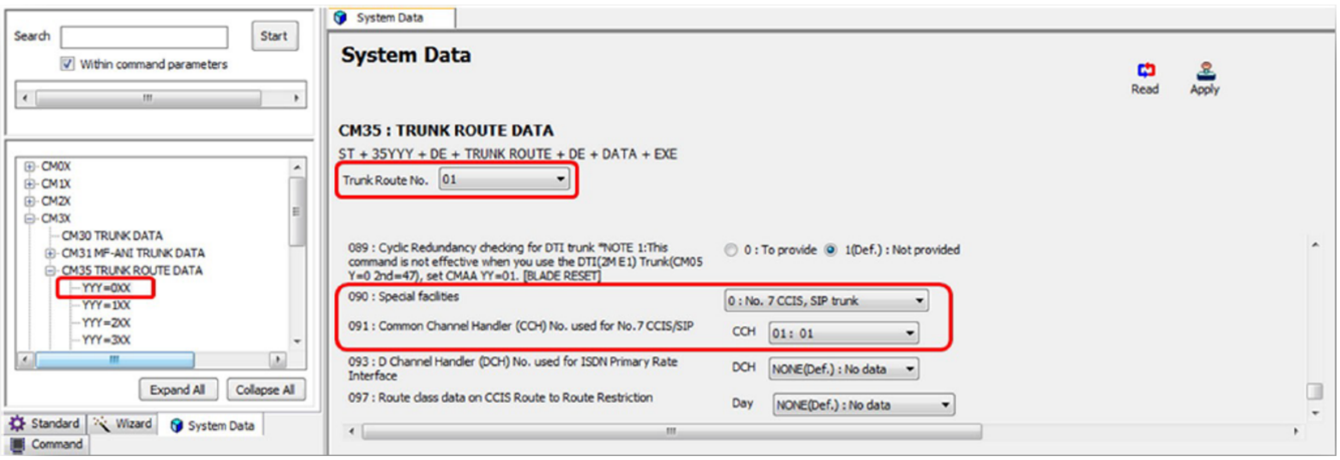

Figure 37 CM 35 SIP Trunk Route Settings (Continued)

1. Select the Trunk Route Number assigned to voice channels and click the **Read** button to get current data settings.
2. YYY=090 Select 0 for SIP trunk service.

   YYY=091 Select CCH used for SIP trunk.

   CCH 00 should not be assigned for SIP trunk; P-P CCIS must used CCH 00.

   Assign a different CCH to each SIP trunk server voice route.
3. Click **Apply**.

Command Line Example:

CM 35090>01>0	(SIP Trunk Route Data: Provide SIP Service)
CM 35091>01>01	(SIP Trunk Route Data: Assign CCH Number)

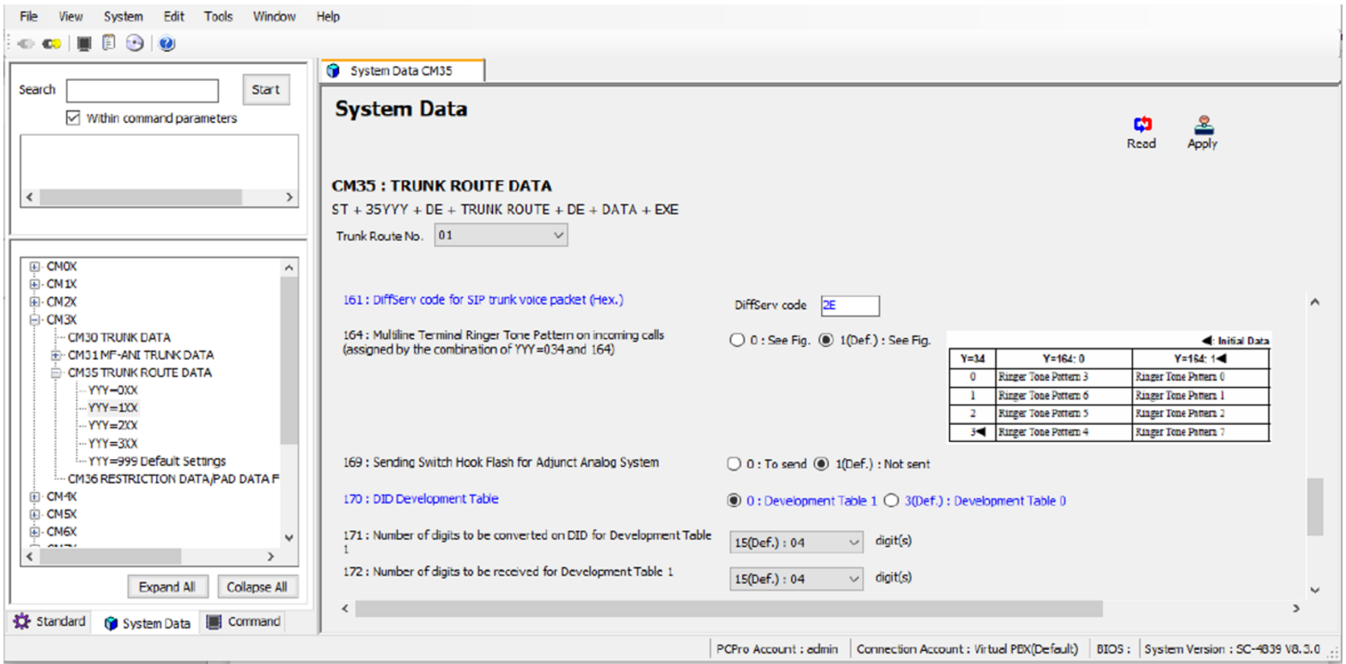

Figure 38 CM 35 SIP Trunk Route Settings (Continued)

1. Select the Trunk Route Number assigned to voice channels and click the **Read** button to get current data settings.
2. YYY=161 Enter DiffServ code in hexadecimal for SIP trunk voice packets (RTP stream).

   Example: 2E = Expedited Forwarding

   YYY=170 set to 0 if receiving more than 4 DID digits

   YYY=171 number of digits to be converted

   YYY=172 number of digits received
3. Click **Apply**.

Command Line Example:

CM 35161>01>2E	(SIP Trunk Route Data: SIP Control QoS DiffServ EF)
CM 35170>01>0	(Development Table 1)
CM 35171>01>15	(Default 04 Digits)
CM 35172>01>15	(Default 04 Digits)

2.9 Route to Route Connection Settings

All values shown are for example purposes only. Your actual values will be determined by your implementation team.

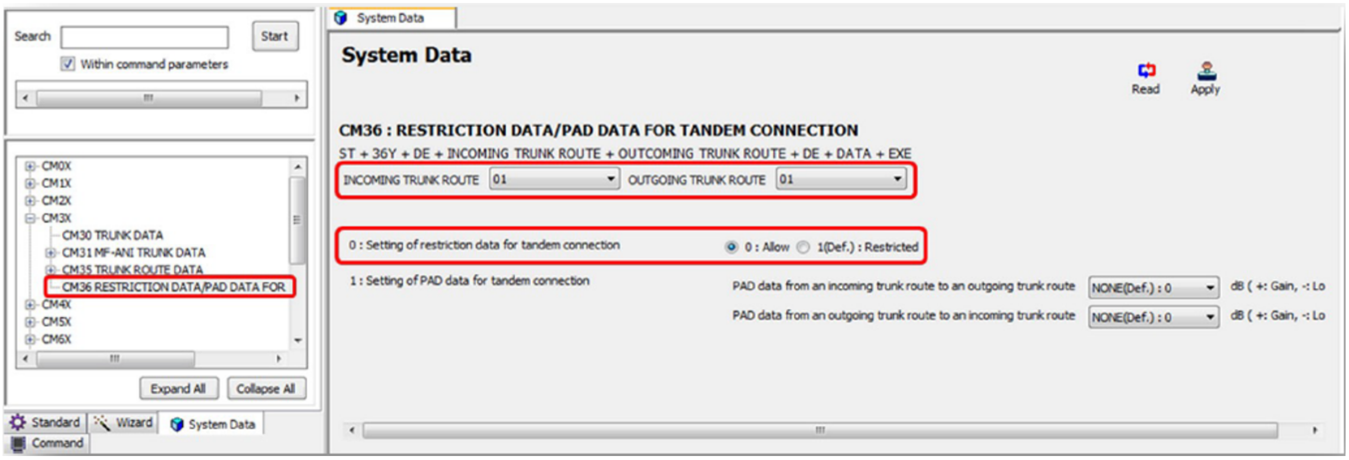

Figure 39 CM 36 SIP Route to Route Connection Settings

1. Select Incoming Trunk Route.
2. Select Outgoing Trunk Route and click the **Read** button to get current data settings.
3. Select 0 to allow route to route connection.

   Allow route to route connection (i.e. SIP route to PRI route, PRI route to SIP route, and SIP route to SIP route).
4. Click **Apply**.

Command Line Example:

CM 360>0101>0 (Route to Route Connection)

2.10 SIP Trunk Control Channel Settings

All values shown are for example purposes only. Your actual values will be determined by your implementation team.

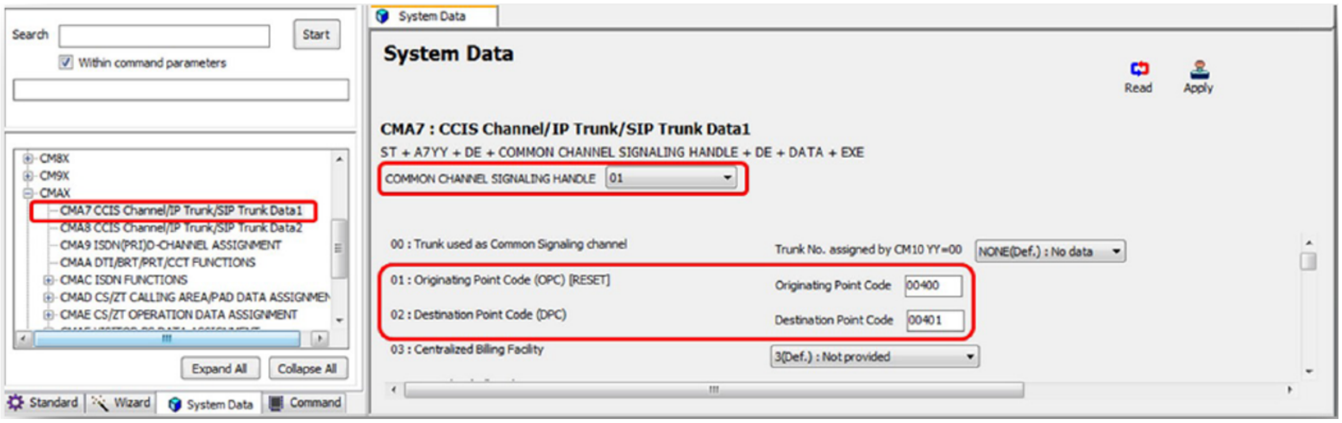

Figure 40 CM A7 SIP Trunk Control Channel Settings

1. Select CCH assigned to SIP trunk and click the **Read** button to get current data settings.

   YY=01 Assign an arbitrary Originating Point Code.

2. The same originating point code can be used for each SIP trunk server.

   YY=02 Assign an arbitrary Destination Point Code.

3. Different destination point code must be used for each SIP trunk server.

4. Click **Apply**.

Command Line Example:

CM A701>01>00400	(Arbitrary Origination Point Code)
CM A702>01>00401	(Arbitrary Destination Point Code)

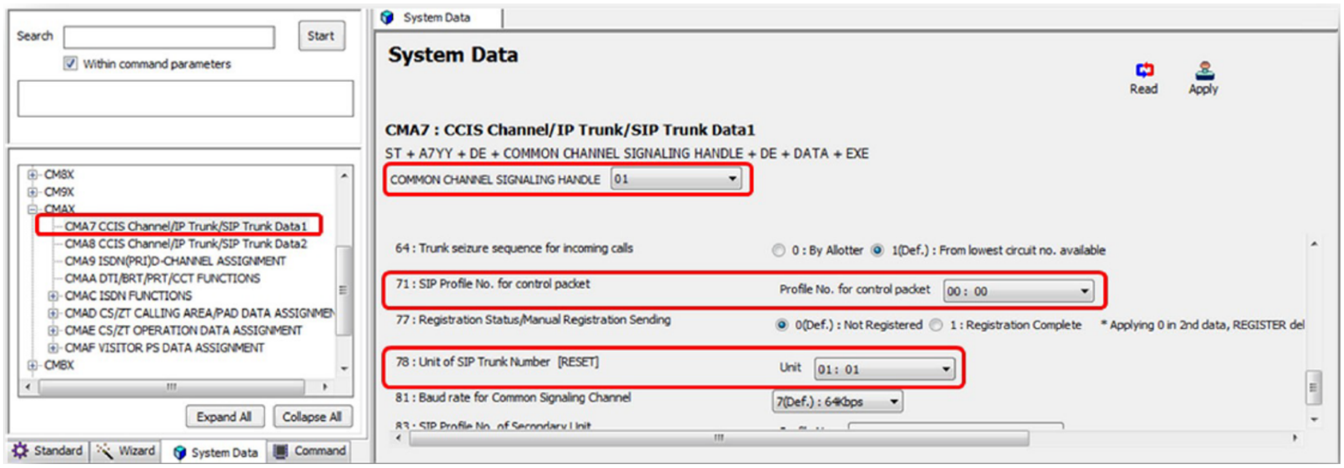

Figure 41 CM A7 SIP Trunk Control Channel Settings (Continued)

1. Select CCH assigned to SIP trunk and click the **Read** button to get current data settings.
2. YY=71 Select an unused SIP Trunk Profile Number.
3. YY=78 Select Unit No. accommodating SIP trunk channels.
4. Click **Apply**.

Command Line Example:

CM A771>01>00	(SIP Trunk Profile 00)
CM A778>01>01	(Main Site Unit 01)

2.11 SIP Trunk Destination Point Code Settings

All values shown are for example purposes only. Your actual values will be determined by your implementation team.

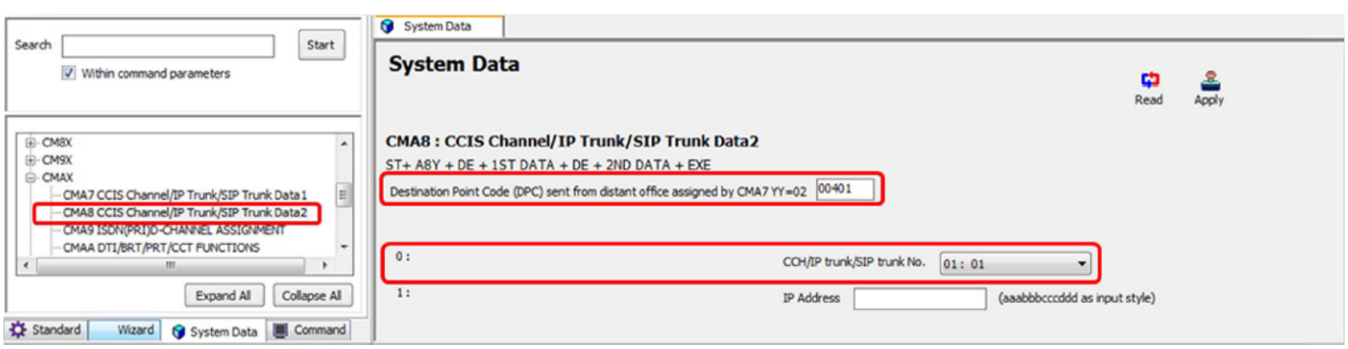

Figure 42 CM A8 SIP Trunk Destination Point Code Settings

1. Enter Destination Point Code assigned by CMA7 YY=02 and click the **Read** button to get current data settings.
2. Y=0 Select CCH assigned to destination point code in CMA7 YY=02.
3. Click **Apply**.

Command Line Example:

CM A80>00401>01 (Destination Point Code 00401 & CCH 01)

2.12 SIP Trunk Profile Settings

Refer to the Tie Trunk management menu in the UNIVERGE BLUE CONNECT Admin Control Panel for the Trunk registration info required for the steps below.

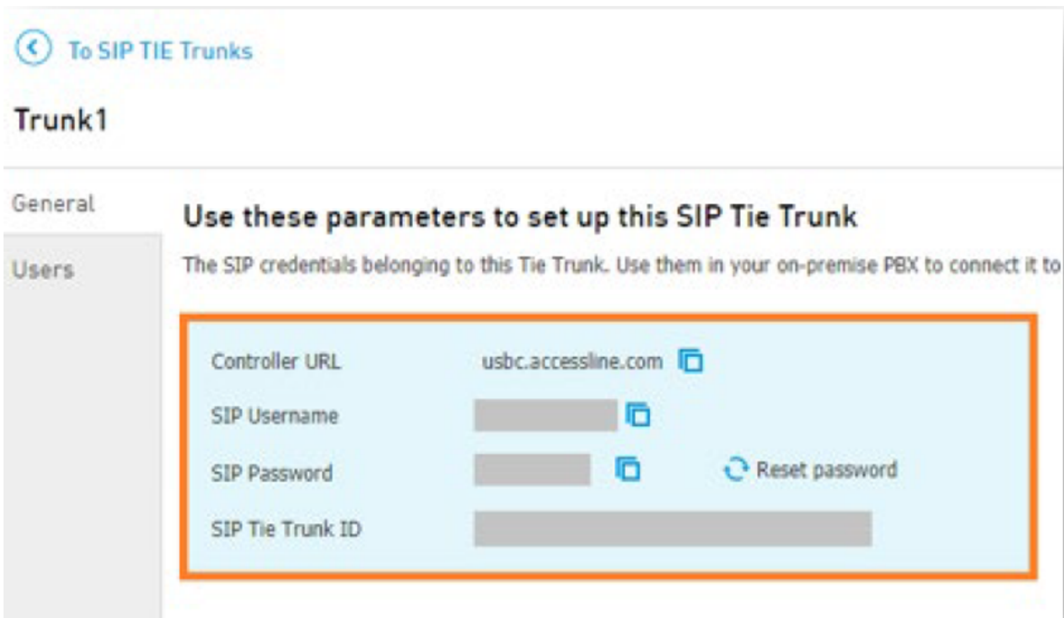

Figure 43 SIP Trunk Profile Settings

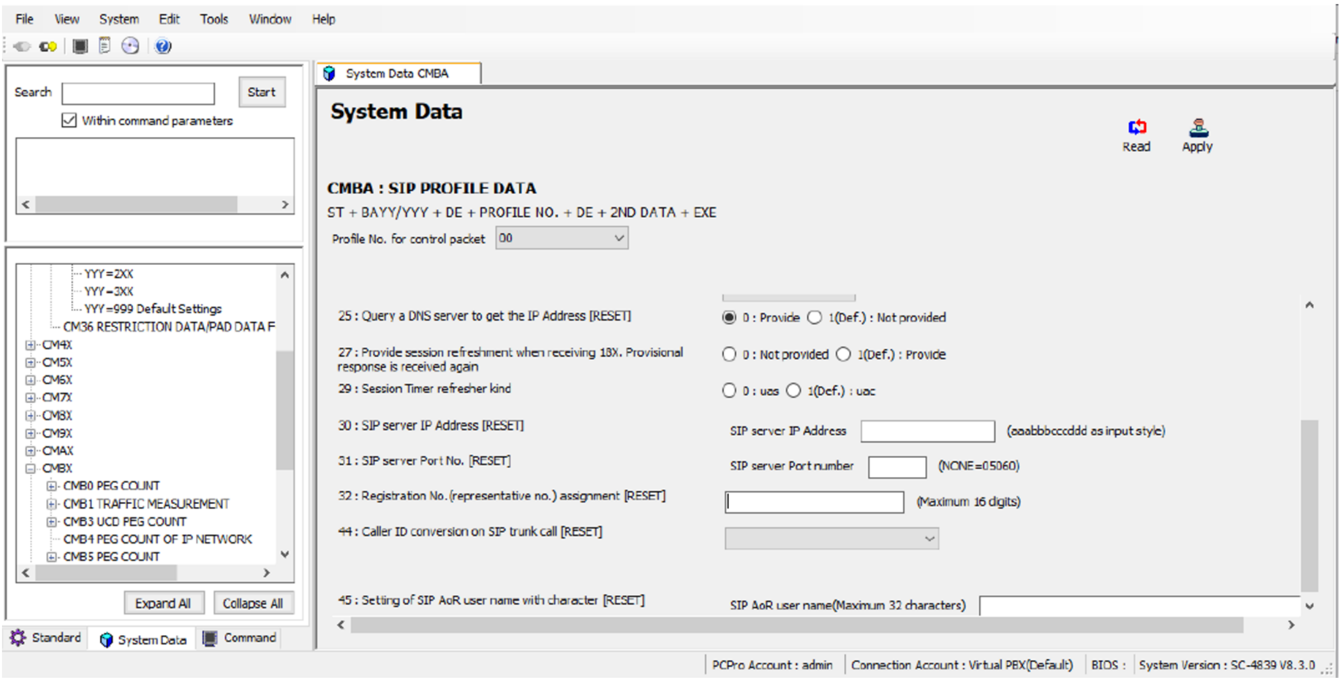

Figure 44 CM BA SIP Trunk Profile Settings (Continued)

1. Select SIP Trunk Profile Number assigned by CM35 YYY=091 and click the **Read** button to get current data settings.
2. YY=25 (Query a DNS server to get the IP Address)
3. YY=32 (Must have a phone number here)
4. YY=45 Enter SIP account Username.
5. Click **Apply**.

Account Name SV9300

CM BA25>00>0	(Query a DNS server to get the IP Address)
CM BA32>00>	(Main billing number 10 digits)
CM BA45>00>SV9300	(SIP Account User Name)

Figure 45 CM BA SIP Trunk Profile Settings (Continued)

1. Select SIP Trunk Profile Number assigned by CM35 YYY=091 and click the **Read** button to get current data settings.
2. YY=52 Select 03 for out of band DTMF (RFC2833).
3. YY=70 Select SIP registration method.
4. YY=72 Enter SIP trunk account Username from the Tie Trunk management menu in UNIVERGE BLUE CONNECT Admin Control Panel.
5. YY=74 Enter SIP trunk account password from the Tie Trunk management menu in UNIVERGE BLUE CONNECT Admin Control Panel.
6. YY=76 Enter SIP trunk domain name when querying DNS server. (use **Controller URL** from the Tie Trunk management menu in the UNIVERGE BLUE CONNECT Admin Control Panel).
7. Click **Apply**.

Command Line Example:

CM BA52>00>03	(Out of Band DTMF RFC2833)
CM BA55>00>7(Def)	(No Identity Header)
CM BA70>00>0	(Registration Required)
CM BA72>00>SV9300	(SIP Trunk Account User Name)
CM BA74>00>Password	(SIP Trunk Account Password)
CM BA76>00>Domain	(Server Domain Name)

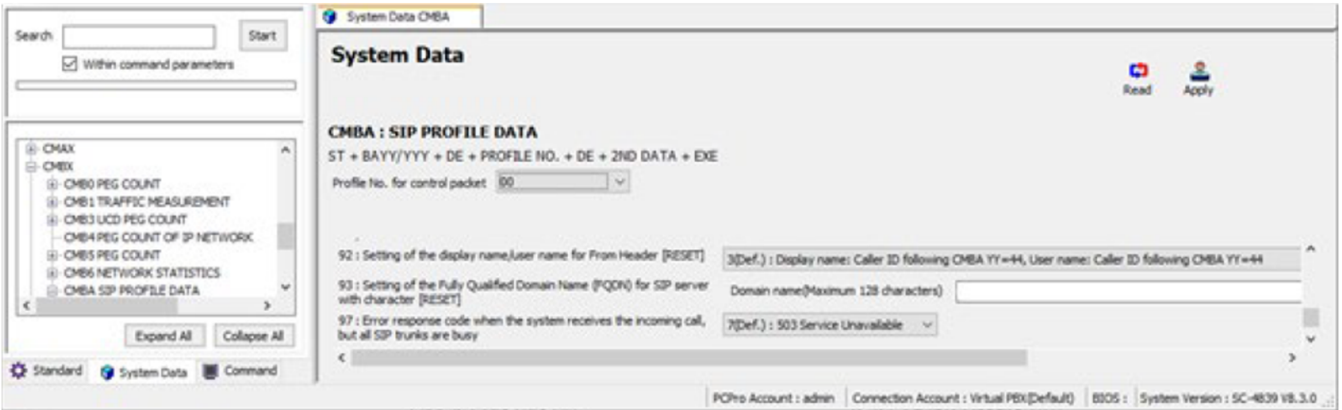

Figure 46 CM BA SIP Trunk Profile Settings (Continued)

1. Select SIP Trunk Profile Number assigned by CM35 YYY=091 and click the **Read** button to get current data settings.

   YY=93 Enter SIP trunk account domain when querying DNS server. (use **Controller URL** from the Tie Trunk management menu in the UNIVERGE BLUE CONNECT Admin Control Panel).

   YY=97 Select response message when all trunks are busy.

2. Click **Apply**.

Command Line Example:

CM BA93>00>2	(domain when querying DNS server)
CM BA97>00>1	(SIP response when all trunks are busy: 488 Busy Here)

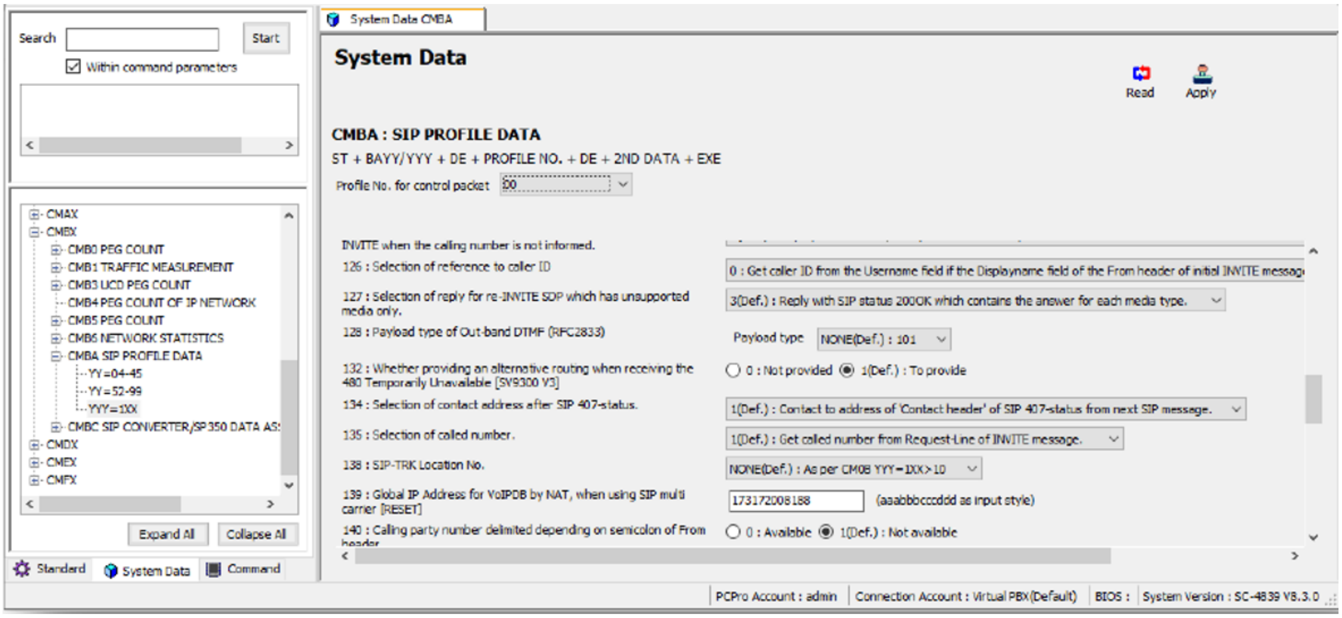

Figure 47 CM BA SIP Trunk Profile Settings (Continued)

1. Select SIP Trunk Profile Number assigned by CM35 YYY=091 and click the **Read** button to get current data settings.
2. YY=126 Select 0 to receive CID from the Username field if the display name field is blank in the initial invite.
3. Click **Apply**.

Command Line Example:

CM BA126>00>0 (Receiving Caller ID Reference)

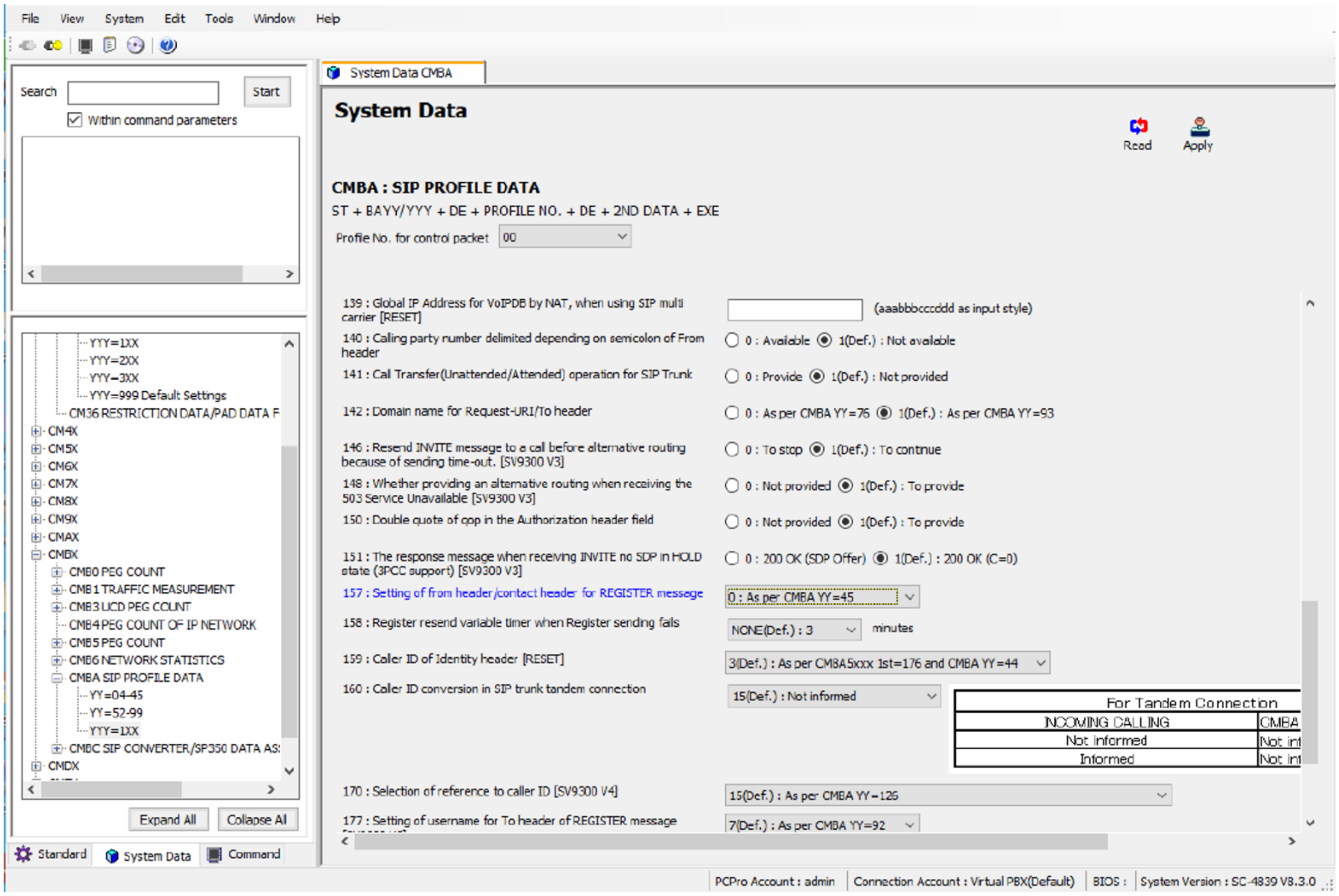

Figure 48 CM BA SIP Trunk Profile Settings (Continued)

1. Select SIP Trunk Profile Number assigned by CM35 YYY=091 and click the **Read** button to get current data settings.
2. YY=139 Enter global IP address when more than 1 SIP trunk profile is used.

   If only one SIP trunk profile is used, enter global IP Address in CM0B YY=101>70.
3. YY=157 Setting of from header/contact header for REGISTER message.
4. YY=160 Select Caller ID format for SIP trunk tandem connection.
5. Click **Apply**.

Command Line Example:

CM BA139>00>066137132001	(Global IP Address)
CM BA157>00>0	(Setting of from header/contact header for REGISTER message)
CM BA160>00>01	(Tandem Conversion Mode 1)

2.13 DID Digit Conversion Settings

All values shown are for example purposes only. Your actual values will be determined by your implementation team.

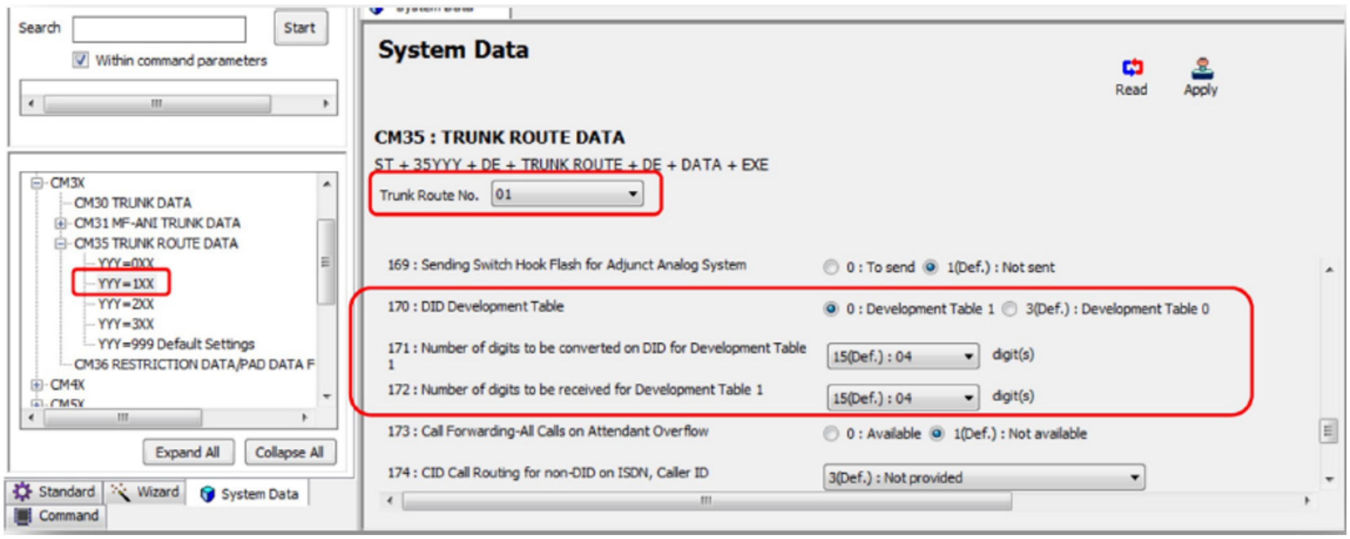

Figure 49 CM 35 DID Digit Conversion Settings

1. Select the Trunk Route Number assigned to voice channels and click the Read button to get current data settings.

   YYY=170 Select 0 to use DID Development Table 1.

   YYY=171 Select the number of digits to be converted. YYY=172 Select the number of digits received.

2. Click **Apply**.

Command Line Example:

CM 35170>01>0	(DID Development Table 1)
CM 35171>01>15	(Default: 04 Digits)
CM 35172>01>15	(Default: 04 Digits)

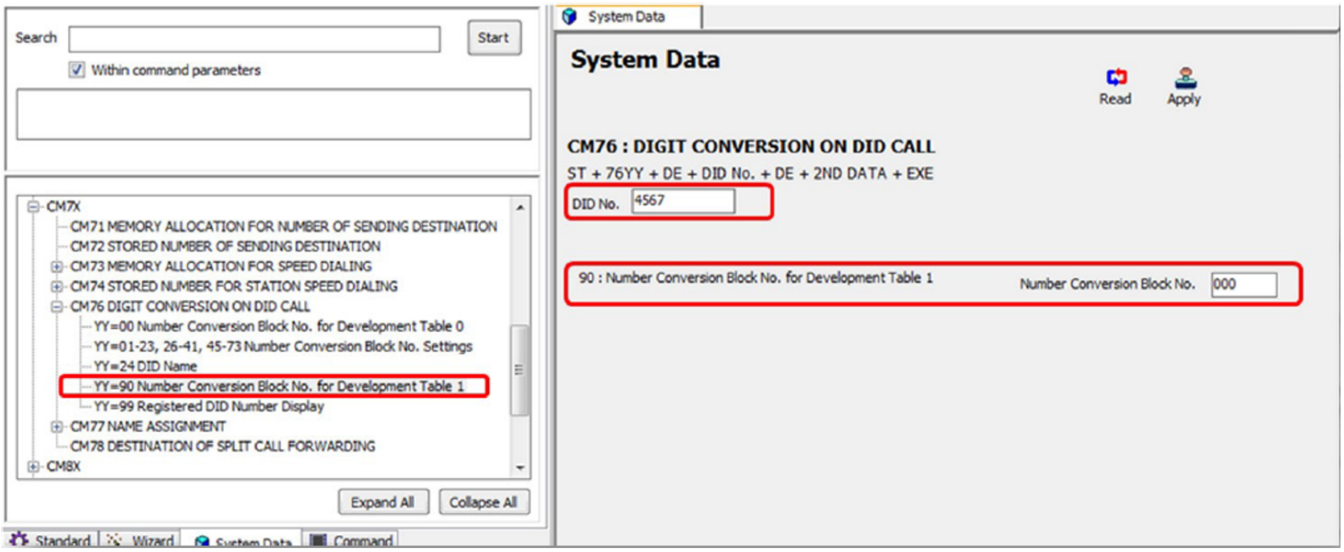

Figure 50 CM 76 DID Digit Conversion Settings (Continued)

1. Enter DID digits to be converted and click the **Read** button to get current data settings.
2. Enter DID conversion block number.
3. Click **Apply**.

Command Line Example:

CM 7690>4567>000 (DID 214-555-4567/Conversion Block 000)

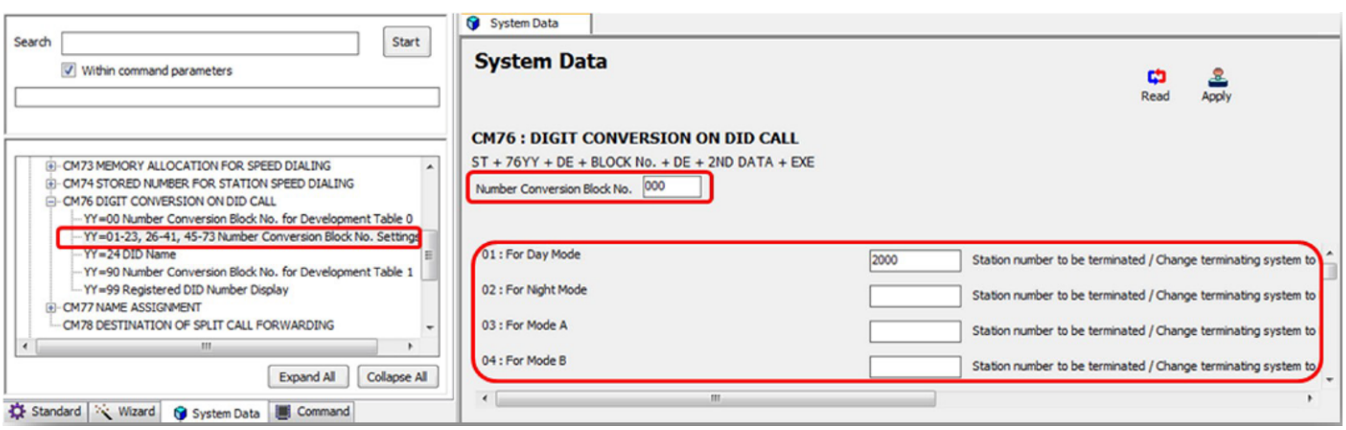

Figure 51 CM 76 DID Digit Conversion Settings (Continued)

1. Enter DID conversion block number and click the **Read** button to get current data settings.
2. YY=01~04 Enter DID destination for the required system modes.
3. Click **Apply**.

Command Line Example:

CM 7601>0000>2000 (Conversion Block 000/Destination Stn. 2000)

2.14 Caller ID Display

All values shown are for example purposes only. Your actual values will be determined by your implementation team.

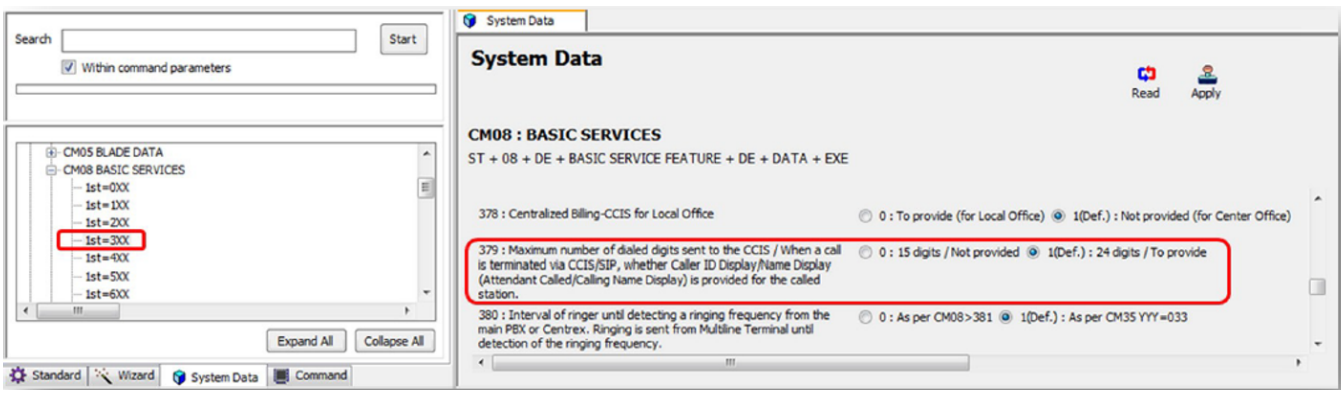

Figure 52 CM 08 Caller ID Display

1. Click the **Read** button to get the current settings.
2. FD=379 Select 1 to provide Caller ID display.
3. Click **Apply**.

Command Line Example:

CM 08>379>1 (Default: Provide Caller ID Display)

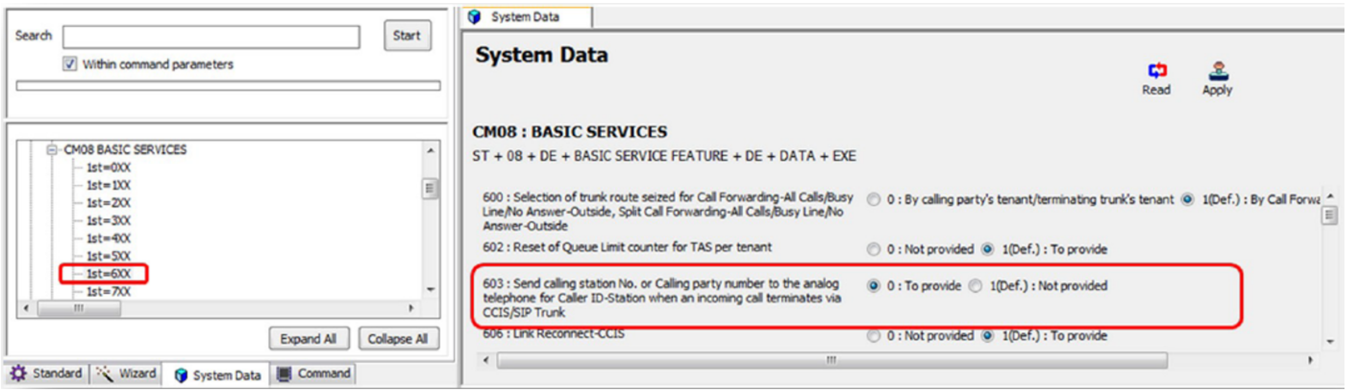

Figure 53 CM 08 Caller ID Display (Continued)

1. Click the **Read** button to get the current settings.
2. FD=603 Select 0 to provide Caller ID display.
3. Click **Apply**.

Command Line Example:

CM 08>603>0 (Caller ID Display on Analog Terminal)

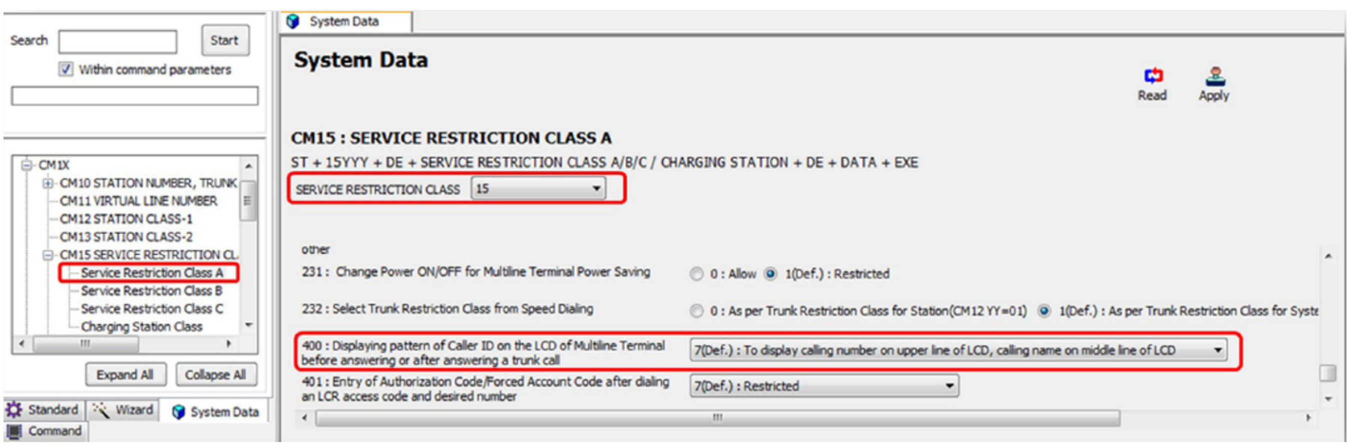

Figure 54 CM 15 Caller ID Display (Continued)

1. Select station Service Restriction Class A and click **Read** to get the current data settings.
2. YYY=400 Select 1 or 7 to display both name and number simultaneously.
3. Click **Apply**.

Command Line Example:

CM 15400>15>7 (Default: Caller ID Number on Upper Line/Name on Middle Line)

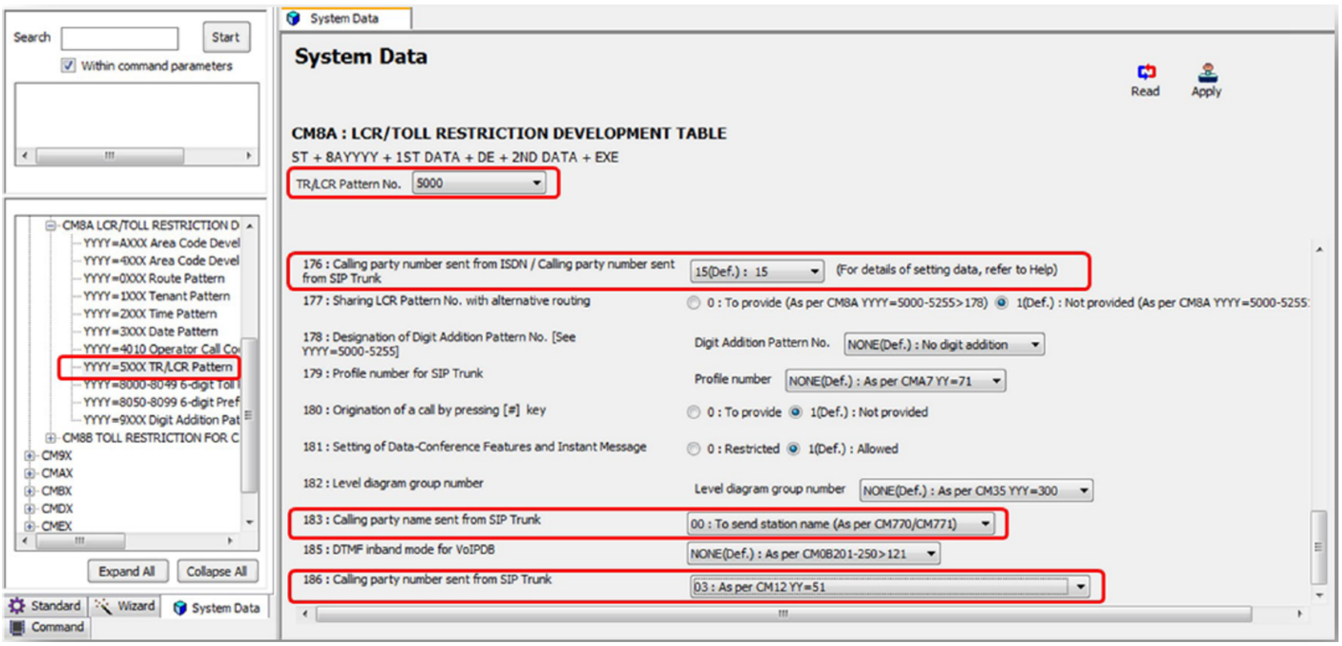

Figure 55 CM 8A Caller ID Display (Continued)

1. Select station SIP Trunk LCR Pattern No. and click **Read** to get the current data settings.
2. FD=176 Select 15 to follow CM 8A5xxx>186 setting.
3. FD=183 Select 00 to send calling party name.
4. FD=186 Select calling party number sending option.
5. Click **Apply**.

command Line Example:

CM 8A5000>176>15	(Default: Follow CM 8Axxx>186 Setting)
CM 8A5000>183>00	(Send Calling Party Name Assigned to CM77 Y=1)
CM 8A5000>186>03	(Send Calling Party Number Assigned to CM12 YY=51)

SECTION 3 INITIAL TESTING AND TROUBLESHOOTING

To confirm that the system is correctly set, perform the following tests:

If you run into an issue with any of these tests, refer to the Table 2 Troubleshooting Guide on the next page.

1. Test an outgoing call to a local number. Check for ringback, 2-way audio and quality.
2. Test an outgoing call to a long distance number. Check for ringback, 2-way audio and quality.
3. Test an outgoing call to an international number. Check for ringback, 2-way audio and quality.
4. Test an outgoing call lasting more than 15 minutes.
5. Test multiple call concurrences on outgoing calls. Setup multiple calls to PSTN.
6. Test an outgoing call to an Operator ‘0’.
7. Test an outgoing call to directory assistance **411**.
8. Test a 911 call.

   Identify to the operator that this is a TEST!
9. Test an incoming call to an internal DID. Check for ringback, 2-way audio and quality.
10. Test an incoming call to an auto-attendant. Check DTMF and audio quality.
11. Test transferring calls off-site.
12. Test an outgoing call to an auto-attendant and verify DTMF.

Table 2 Troubleshooting Guide

Issue	Cause	Remedy
No Calls IN/Out	❍ Router Configuration	❍ Check Router Configuration
	❍ NEC Configuration	❍ Check NEC Configuration
	❍ Unqualified IP Address	❍ Note WAN IP Address and Contact Provider
No Calls Out	❍ NEC Configuration	❍ Check NEC Configuration
	❍ Unqualified IP Address	❍ Note WAN IP Address and Contact Provider
No Calls In	❍ NEC Configuration	❍ Check NEC Configuration
	❍ Unqualified IP Address	❍ Note WAN IP Address and Contact Provider
One-Way Audio	❍ NEC Configuration	❍ Check NEC Configuration
Echo	❍ Excessive Delay	❍ Check LAN and WAN for high latency
	❍ Echo Cancellation Issue	❍ Check Echo settings and/or consult SIP trunk provider.
Call Dropping	❍ Internet Access Issues	❍ Call Internet Access Provider
	❍ Extreme Latency on LAN	❍ Check Latency on LAN
	❍ SIP issue	❍ Contact Provider
Static or HUM on Phones	❍ Power issue	❍ Check power if using AC, should not be issue in PoE
Missing Parts of Words	❍ Packet Loss or Latency on LAN	❍ Check LAN
	❍ Packet Loss or Latency on WAN	❍ Check with Internet Access Provider
	❍ Jitter Buffer Configuration	❍ Check with NEC

Command Line Example:

CM 0B101>00>172.24.142.55	(Unit 01 VoIP Address)
CM 0B101>01>255.255.255.0	(Unit 01 VoIP Subnet)
CM 0B101>02>172.24.142.1	(Unit 01 VoIP Default GW)
CM 0B101>10>NONE	(Default: Location 00)
CM 0B101>36>NONE	(Default: SIP Trunk Port 5060)
CM 0B101>60>151.164.1.8	(Unit 01 Primary DNS Server Address)
CM 0B101>61>151.164.11.201	(Unit 01 Secondary DNS Server Address)
CM 0B101>70>66.137.132.162	(Unit 01 Global IP Address)
CM 0B201>00>172.24.142.56	(Unit 01 VoIP IPPAD Address)
CM 0B101>10>32	(Unit 01 VoIP IPPAD Channels)
CM 0B101>50>1	(Unit 01 Allow Fax over IP)
CM 1003>01000>D001	(Unit 01 SIP Trunk Channel 1)
CM 1003>01001>D002	(Unit 01 SIP Trunk Channel 2)
CM 1003>01002>D003	(Unit 01 SIP Trunk Channel 3)
CM 1003>01003>D004	(Unit 01 SIP Trunk Channel 4)
CM 3000>001>01	(Voice Trunk Route Number 01)
CM 3000>002>01	(Voice Trunk Route Number 01)
CM 3000>003>01	(Voice Trunk Route Number 01)
CM 3000>004>01	(Voice Trunk Route Number 01)
CM 3035>001>001	(Assign CIC to Channel 1)
CM 3035>002>002	(Assign CIC to Channel 2)
CM 3035>003>003	(Assign CIC to Channel 3)
CM 3035>004>004	(Assign CIC to Channel 4)
CM 35000>01>00	(SIP Trunk Route Data: Provide SIP Service)
CM 35004>01>2	(SIP Trunk Route Data: Provide SIP Service)
CM 35009>01>03	(SIP Trunk Route Data: Provide SIP Service)
CM 35018>01>0	(SIP Trunk Route Data: Provide DID Digit Conversion)
CM 35020>01>00	(SIP Trunk Route Data: Provide SIP Service)
CM 35090>01>0	(SIP Trunk Route Data: Provide SIP Service)
CM 35091>01>01	(SIP Trunk Route Data: Assign CCH Number)
CM 35161>01>2E	(SIP Trunk Route Data: SIP Control QoS DiffServ EF)
CM 35170>01>0	(DID Development Table 1)
CM 35171>01>15	(Default: 04 Digits)
CM 35172>01>15	(Default: 04 Digits)
CM 360>0101>0	(Route to Route Connection)
CM A701>01>00400	(Arbitrary Origination Point Code)
CM A702>01>00401	(Arbitrary Destination Point Code)
CM A771>01>00	(Assign SIP Trunk Profile Number)
CM A778>01>01	(Assign SV9300 Unit Number with SIP Trunk Channels)
CM A80>00401>01	(Destination Point Code 00401 & CCH 01)
CM BA25>00>0	(Query a DNS server to get the IP Address)
CM BA30>00>	(SIP server IP address)
CM BA31>00>05060	(SIP server port)
CM BA32>00>	(Must have a phone number in this command)
CM BA45>00>	(SIP Account User Name)
CM BA52>00>03	(Out of Band DTMF RFC2833)
CM BA70>00>0	(Registration Required)
CM BA72>00>	(SIP trunk account user name)
CM BA74>00>Password	(SIP trunk account password)
CM BA76>00>Domain	(Server Domain Name)

CM BA93>00>2	(Domain when querying DNS server same as BA76)
CM BA97>00>1	(SIP response when all trunks are busy: 488 Busy Here)
CM BA126>00>0	(Receiving Caller ID Reference)
CM BA139>00>	(Global IP Address/WAN for NAT)
CM BA157>00>0	(Setting of the form header for Register message)
CM BA160>00>01	(Tandem Conversion Mode 1)
CM 7690>4567>000	(DID 214-555-4567/Conversion Block 000)
CM 7601>000>000	(Conversion Block 000/Destination Stn.4XX)
CM 08>379>1	(Default: Provide Caller ID Display)
CM 08>603>0	(Caller ID Display on Analog Terminal)
CM 15400>15>7	(Default: Caller ID Number on Upper Line/ Name on Middle Line)
CM 8A5000>176>15	(Default: Follow CM 8A5xxx>186 Setting)
CM 8A5000>183>00	(Send Calling Party Name Assigned To CM77 Y=1)
CM 8A5000>186>03	(Send Calling Party Number Assigned to CM12 YY=51)

To setup Mobility Access do the following:

Mobility Access will require the following commands to ring the desk and cloud terminal at the same time:

A mobility access license is required for each cloud extension used in the system.

This is just an example; my test bed is using the following:

Trunk route 01 for sip tie lines from the above programming.

Cloud terminals are 4XX series extensions.

Desk extensions are 5XX series.

Access code of 8 to get to the sip tie line route.

CM 08>1026>0 station base

CM 200>8>A127	(LCR Group 1)
CM 8AA000>1>4006	(Area Code Development Pattern No 6)
CM 8A4006>4>0000	(Route Pattern 000)
CM 8A0000>1>00001	(Route Pattern 000/Trunk Route)
CM 856>4>04	(Max digits dialed)

CM 6410>01>8	(Tenant 01/ Trunk access code for Call Forwarding in Mobility Access mode)
CME650>5XX>4XX	(Link desk extension to cloud extension)
CM 1277>5XX>0	(Desk extension turn on Dual Ringing)
CM 1288>5XX>0	(Mobility Access available)
CM 410>162>02	(Dual Ring start timer set to 2 seconds)

CONNECT BRIDGE with PBX Integration

SECTION 1 SUPPORTED FUNCTIONALITY

Account Setup

1. Control Panel UI to create Tie trunks

Tie Trunk Configuration

1. BRIDGE integrates with NEC PBX via a SIP Tie Trunk
2. BRIDGE allows multiple PBX to be connected to the same account.
3. Tie Trunk allows assigning of Users, which converts them into Trunk Users.
4. Tie Trunk allows programming of Cloud Extension, which is used by the NEC PBX to route inbound calls to the User over the Tie Trunk.
5. Tie Trunk allows programming of Premise Extensions, which are non-User dialable extensions on the NEC PBX.
6. Tie Trunk supports Codec priority selection.
7. Tie Trunk supports up to 1000 concurrent phone calls.
8. Tie Trunk supports addition of a trunk access code to outbound PSTN calls.
9. Tie Trunk allows configuration of Trunk User outbound PSTN caller ID.

Dialing

1. BRIDGE allows Ext to Ext dialing:
   a Between Users on the BRIDGE account
   b Between BRIDGE and PBX Users
   c Between PBX users
2. BRIDGE allows PSTN dialing via the PBX’s PSTN connection:
   a Pure PBX Users dial PSTN numbers over the PBX’s PSTN connection
   b Trunk Users dial PSTN numbers over the PBX’s PSTN connection

3. BRIDGE supports the following countries / Dial Plans:
    a North America (NANP)
    b UK
    c NL
    d DE
    e IT
    f AUS
    g JP
4. BRIDGE supports 3 to 6-digit extension ranges.

Calling Features

1. BRIDGE supports the use of Call Forwarding and can route forwarded calls.
2. BRIDGE supports the use of Conferencing and can route conferenced calls.
3. BRIDGE supports the use of Park between UNIVERGE BLUE desktop and mobile applications.
4. BRIDGE supports the use of Park between NEC terminals.
5. BRIDGE supports voicemails on the UNIVERGE BLUE side of the solution
6. BRIDGE supports the sending of caller ID, and will send an extension number for internal calls, and an E.164 formatted phone number for outbound PSTN calls.
7. BRIDGE supports passing caller ID name (CNAM) on calls.
8. BRIDGE supports Call Flip between the UB applications.
9. BRIDGE supports Call History on the NEC terminal, and in the UB applications.

Desktop Application

1. BRIDGE allows Trunk Users to make and receive calls to NEC PBX extensions over the Tie Trunk.
2. BRIDGE allows Trunk Users to dial PSTN numbers over the Tie Trunk and over the NEC PBX connection.

Mobile Application

1. BRIDGE allows Trunk Users to make and receive calls to NEC PBX extensions over the Tie Trunk.

2. BRIDGE allows Trunk Users to dial PSTN numbers over the Tie Trunk and over the NEC PBX connection.

SECTION 2 LIMITATIONS

Account Setup

1. 5000 User limit per BRIDGE account.

Tie Trunk Configuration

1. BRIDGE only integrates with 4 NEC PBX currently:
   - a SV9100
   - b SV9300
   - c SV9500
   - d 3C
2. BRIDGE will soon integrate with:
   - a SIP@Net
3. BRIDGE will not currently integrate with:
   - a SL2100
   - b Any other PBX or SIP gateway device

Dialing

1. BRIDGE requires all PBX to be connected to the same account to have the same Ext length.
2. BRIDGE only supports 7 dial plans currently and thus can only be sold in those 7 countries, although many EMEA countries use essentially the same dial plans so there is likely more potential there for expansion.
3. Calling any Trunk User from within the Bridge account (from any of the applications) will result in a **looping** call across the Tie Trunk, where the call that is sent to ring the Trunk User's NEC terminal is sent back across the Tie Trunk, resulting in 2 inbound calls occurring on the Trunk User's applications.

   NOTE: This limitation is removed from SV9300 system software v9.2.1 or higher.
4. Bridge only supports 3-6-digit extensions. PBX requiring 7 or 8-digit extensions cannot currently be supported.

5. If the NEC PBX terminal is offline, inbound calls fail to a busy signal and will not ring on the UB applications.

6. If both UB applications are offline, and the User has never signed into the Mobile application, inbound calls will go immediately to voicemail on the Bridge side.

7. * and # codes dialed within the UB applications will not send across the Tie Trunk to the PBX, so features such as Call Forwarding can't be enabled or disabled on the PBX side from the UB applications.

Calling Features

1. Bridge currently cannot park and retrieve phone calls across the Tie Trunk between platforms.

2. Bridge currently requires voicemail to be disabled on the NEC PBX.

3. Some PBX (SV9500) require additional programming to handle being sent a PSTN phone number instead of an extension number on outbound calls to the PSTN.

4. Inbound calls will fail if the NEC PBX terminal (on the SV9100 at least) if the terminal is not actively registered. Calls will not proceed across the Tie Trunk to the applications.

5. Inbound calls will go immediately to voicemail if the Trunk User does not have at least one UB application actively registered. The call may ring on the PBX terminal very briefly but that's it.

6. When the remote side of a call with a Trunk User hangs up first, it's common (on the SV9100) to hear dial tone. This also happens when the remote side leaves a voicemail. The voicemail ends with 30 seconds of dial tone.

7. * and # codes do not send from the Bridge account across the Tie Trunk, as the UB HPBX has its own * and # codes, and upon detection, attempts to determine what to do with those commands.

8. Call Flip does not work across the Tie Trunk from the applications to the NEC terminal, and vice versa.

9. Call History does not record calls occurring on the other side of the Tie Trunk. Calls placed from the UB applications will not register a call in the history on the NEC terminal. Calls placed from the NEC terminal will not register a call in the history on the UB applications.

10. Some PBX (at least SV9100) do not support sending CNAM across the Tie Trunk, so no CNAM appears on inbound phone calls in the applications.

Desktop Application

1. Calls made from the NEC terminal will not display in the applications' call history.

Mobile Application

1. Calls made from the NEC terminal will not display in the applications' call history.