

Ser Guide

April 2025

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Welcome to SIP Trunking

SIP, or Session Initiation Protocol, has become the standard voice connection method in the VoIP industry. A SIP trunk is a connection that carries calls for multiple endpoints, as opposed to most other SIP devices which carry calls for only a single endpoint. Trunks are usually connected to an on-premises PBX (private branch exchange) and are typically billed on a per-concurrent call basis in addition to any toll charges for calls placed across the trunk.

A SIP trunk can be set up to support any number of concurrent calls, or channels, but a specific trunk is limited to the concurrent call limit. Once that limit is reached, incoming calls follow the Overflow Routing settings. If a trunk fails, incoming calls follow the Inbound Failover Routing settings. Multiple SIP trunks allow for calls to failover to associated trunks seamlessly, allowing your service to continue if something happens to prevent calls in one office.

Access the Voice Portal

The Voice Portal is where you can manage the SIP trunks and settings on your account. To add a new SIP trunk, however, please contact Customer Support.

When your account was created, you were sent an email that contains your username, a link to create your password, and the portal URL. Keep that email safe so you can refer to it later. If you haven't received it, or if you don't have an email address on your account yet, please contact Customer Support.

- 1. Go to https://user.cymbus.com.
- 2. Enter your username (not email) and password.
- 3. Check *Remember Me* if you want to save your username and password.
- 4. Click Login.

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Login	
Email A	ddress or Username*
	Forgot Password?
Passwo	rd*
Ren	ember Me



Password Requirements

Your password must be at least eight characters long and include at least one number and one special character, such as $\$ * .[] \{ \} () ? "! @ # % & / \, > < ' : ; |_~ ~ ` = + -. It cannot contain any part of your username.$

Reset Your Password

If you can't log in, you can reset your password without contacting Customer Support.

- 1. On the login screen, click the Forgot Password? link.
- 2. Enter the email address associated with your account, then click **[Reset Password]**. If you don't have an email address on your account yet, please contact Customer Support for assistance.
- **3.** Open the email and click the link. If it opens on the login screen, click the link again to go to the right place.
- 4. Enter a new password that is at least 8 characters long and includes at least one number and one special character, such as (^ \$ * . [] { } () ? "! @ # % & / \ , <> ' : ; | _ ~ ` = + -).

Create Pa	ssword d*		
•••••			Ο
Includes a	a minimum of 8 ch	aracters	
Includes a	at least one numbe	er	
Includes a	at least one specia	l character	
	Save Passw	ord	

5. Click [Save]. A confirmation email will be sent to your email address.

Language Setting

Experience the Alianza platform in English, Canadian French, or Spanish, seamlessly integrated across audio prompts, system-generated emails, and more.

All users can change the language for their current session from the menu in the top right corner of the screen. Additionally, Admin end users can also set a default language for the account and each SIP trunk.

Change Session Language

Follow the steps below to change the language used in the web portal for your active session:

- 1. Log in to the Voice Portal.
- 2. In the top right corner of the page, expand the menu with your name and select Language.
- **3.** Choose a language: English, French, or Spanish.



4. Click **[Save]**. The portal will refresh to update with the selected language.

The portal will revert to your default language once you log out.

Donna Noble Documentation 🔻	Select a Language	Select a Language
Sign Out	Spanish •	English -
Language - English	English	
	French	Cancel Save
	Spanish 🗸	

Change Default Language

A language can be set in the following locations:

- Account. The language setting on an account applies to all users by default. Changing this setting will apply to new users only; existing users will maintain their current setting. See Account Details.
- Account Managers. The user's language setting applies to the web portal, IP phone (if the phone supports the language), telephone user interface (TUI) audio prompts, and system-generated emails. It does not affect the user's voicemail box, which is configured separately. See Account Managers.
- **SIP Trunk.** The language setting on a SIP trunk applies to the telephone user inteface (TUI) audio prompts. It does not affect the language used in the portal. See Manage SIP Trunks.

Manage SIP Trunks

To manage your SIP trunks, log in to the Admin Portal and go to **Account > SIP Trunk**. From here, select an existing trunk to view or modify the settings.

Here you can configure the SIP trunk settings, including forwarding, overflow and failover routing, concurrent call limits, and calling plans.

SIP Trunks				
Name 个	Concurrent Calls Allowed	Extensions	Registration Status	
West Region	80	4000	Registered C	
East Region	120	3000	Unregistered C	

Details

Details	
Trunk Name	
Use IP authentication	
Language	-
Trunk Name	West Region
Use IP authentication	
Registration Status	Yes 🗸
Assigned Phone Numbers	1 (801) 555-9000
911 Callback Number	1 (801) 555-7458 - 100 S 200 W, Orem, UT 84058
Primary Phone Number	None Used for identifying outbound calls from this SIP trunk for billing and local services (not used for caller ID).
Enable Local Services	Route N11 calls to local services (learn more)
Number of Dialing Digits	11-Digits 🔹

Field	Description
Trunk Name	The name of the given SIP trunk.
Use IP authentication	Identifies whether this SIP trunk uses IP authentication. If unchecked, the trunk will authenticate via user-based SIP credentials, which will be provided in the <i>Authentication Credentials</i> section below (once the trunk is saved).
Language	 Select a language for the trunk, which will be used for audio prompts: English French (Canadian) Spanish

Field	Description	
911 Callback Number	The phone number and address that emergency services will be given when a call comes in from the given SIP trunk. All phone numbers that call emergency services via this SIP trunk will have this phone number and address. Only one phone number and address can be assigned to a given SIP trunk.	
Primary Phone Number	Select the phone number which will be used to identify outbound calls from this SIP trunk for billing and local services (not for caller ID).	
Enable Local Services	Choose whether to enable local services dialing. Once enabled, the voice server will accept and route 3-digit N11 service numbers (411, 611, etc.) and 7-digit local dialing numbers from the SIP trunk. See Local Services for details.	
Digits Received	 Select the number of digits the SIP trunk will receive for inbound calls: 7-digits: 123-4567 10-digits: 801-123-4567 11-digits: 1-801-123-4567 	

Account Codes

If Account Codes is enabled for the account, check this box to enable it on the SIP trunk.

Account Codes	
Enable account codes	

Forward All

Forward All		
Forward All	All inbound calls will be forwarded (ignores Failover Routin	g)
	SIP Trunk	•
	East Region	•

Field	Description
Forward All Calls	Check this box to forward all inbound calls intended for this SIP trunk to a specific destination:
	SIP Trunk
	• User
	Auto-Attendant
	Voicemail
	Telephone Number
	Busy Signal

Overflow Routing

When the calls to the SIP trunk reach capacity (it has as many calls as allowed by the concurrent call setting on the SIP trunk), the Alianza platform will attempt to route calls according to this configuration. It will try these items in priority order. Multiple SIP trunks can be tried along with finally routing to another destination if all the attempts to route to configured SIP trunks fail.

In capacity exceeded scenarios, calls being routed to other SIP trunks will **not** follow the other SIP trunk's failover routing configuration.

Overflow Routing		
	+ Add Overflow SIP Trunk	
	When this SIP Trunk reaches capacity, it will attempt to route calls to the first available SIP Trunk, starting at the top of this list.	
Forward on Overflow	Busy Signal 👻	
	If all SIP Trunks reach capacity, calls will be forwarded to this destination.	

Field	Description
[+ Add Overflow SIP Trunks]	Creates a new menu to select which SIP trunk calls will failover to.

Field	Description
Forward on Overflow	If all SIP trunks reach the concurrent call limit, calls will be forwarded to the configured destination:
	• User
	• IVR
	• Voicemail
	Telephone Number
	Busy Signal

Inbound Failover Routing

When the calls to the SIP trunk fail, the Alianza platform will attempt to route calls according to this configuration. It will try these items in priority order. Multiple SIP trunks can be tried along with finally routing to another destination if all the attempts to route to configured SIP trunks fail.

In failover scenarios, calls being routed to other SIP trunks will **not** follow the other SIP trunk's failover routing configuration.

Inbound Failover Routing			
	+ Add Failover SIP Trunk		
	When this SIP Trunk fails, it will attempt to SIP Trunk, starting at the top of this list.	route calls to the first available	
Forward on Failure	Busy Signal	.	
	If all SIP Trunks fail, calls will be forwarded	d to this destination.	

Field	Description
[+ Add Failover SIP Trunks]	Creates a new menu to select which SIP trunk calls will fail over to.

Field	Description
Forward on Failure	If all SIP trunks fail, calls will be forwarded to the configured destination:
	• User
	• IVR
	• Voicemail
	Telephone Number
	• Busy Signal

Extension Dialing

For environments with a blend of telephony technologies, SIP trunk extension dialing will reduce phone costs and improve local communications. It allows you to assign extensions to SIP trunks *and* Business Cloud Communications users on the same account, enabling those extensions to dial each other. This means that:

- A user with an extension on the SIP trunk can dial the extension of an IP phone.
- A user with an IP phone can dial an extension on the SIP trunk.

In one example, we helped a retirement community radically overhaul their ease of dialing and reduce costs. The community has an onsite nursing wing, assisted living apartments, and private retirement cottages that needed to bridge the communication gap between their onsite PBX connected by a SIP trunk and IP phones connected to hosted PBX lines. The SIP trunk extension dialing feature enables the residents and caretakers to reach anyone in the community via their 3-4 digit extensions, regardless of their location and telephone technologies.

Assign Extensions to SIP Trunk

Here's how you do it:

- 1. First, go to **Account > Settings > Devices** and check your settings. For extension dialing to function properly, **Dialing Behavior** must be set to *Open Dialing* or an option that allows an extension.
- 2. Next, go to Account > SIP Trunks > Trunks.
- 3. Select a SIP trunk or add a new one.
- **4.** Under **Definition**, assign extensions to the SIP trunk. You can enter single extensions, variable extensions, or an extension range in a string; separate each value by a comma.
 - **Single:** 2100 = Extension 2100
 - Variable: 21XX = All extensions between 2100 and 2100
 - Range: 2110-2199 = All extensions between 2100 and 2199
 - String: 4200-5500, 5501, 5578, 68XX

Note

A single extension cannot be assigned to multiple locations (user, group, or SIP trunk). If an end user is assigned an extension that overlaps with an extension range assigned to a SIP trunk, calls to that extension will be routed to the *end user*, not to the SIP trunk.

Additionally, if an end user extension overlaps an extension range assigned to a SIP trunk, calls to that extension will be routed to the user and **not** the SIP trunk.



Account Codes

Account Codes, also known as authorization codes, are used to tag (classify) calls for billing and reporting purposes and/or require authorization for calls to international locations, directory assistance, and premium toll numbers.

Admin users will create specific codes that users can dial to associate call activity with a department, project, client account, and more. For example, if an agency needs to keep track of billable client calls, they can dial the code that corresponds to the client, and the code will be tagged in the Call History record. Later, the billing department can review the call records and bill each client appropriately.

Additionally, if an organization wants to require authorization for international calls, they can create codes for each user, then configure the account to require a valid account code for international calls. When a caller places a call to a country code that is different from their own, they will be asked to enter their code to authorize the call.

For additional use cases and how to set them up, see Account Codes Settings.

How to Use Account Codes

The general flow for using an account code on inbound and outbound calls is outlined below. However, it will change depending on which settings are enabled and disabled. See Account Codes Settings for details.

- **Outbound Calls:** Dial a phone number. When prompted, enter an account code (for example, 123).
- **Inbound Calls:** Answer an inbound call, then dial the star code + the account code (for example, *50123). The other party can hear the dial tones, so let them know what you're doing first.

Mid-Call Star Code

If you don't know which star code is assigned to Account Codes, ask Customer Support. The star code allows users to tag inbound calls or outbound calls that don't require an account code.

Account Codes Settings

Account Codes settings can be configured for a variety of use cases. The default settings determine if codes are optional (used for tracking) or required (used for authorization), and the second group of settings determine what type of calls require them.

These settings are configured at the account level. See Account Codes Setup for details.



Default Settings

Require Account Codes for Internal Calls

If this setting is enabled, users are prompted to dial an account code on every outbound call that originates and terminates on this account.

- **On (Required):** The user dials an on-net phone number or extension and hears, "Please enter a valid account code."
- **Off (Optional):** The user dials an on-net phone number or extension. If they want to tag the call, the user can dial the star code + account code once the call is answered.

Validate Account Codes When Dialed

If this setting is enabled, outbound calls are not connected unless the caller enters a valid code. However, an inbound call will continue even if an incorrect code is entered to the max retry limit.

- **On (Validated):** The account code is checked against the codes on the account. Outbound calls are not connected unless the caller enters a valid code. Inbound calls, however, will continue even if the user enters invalid codes up to the max retry limit.
 - If the code is valid, the user hears a splash tone to indicate the account code was accepted. The call continues and the code is added to the Call History record.
 - If the code is not valid, the user hears, "Invalid entry. Please enter an X-digit account code."
- Off (Not Validated): The user can enter any code they want as long as it's the right length.
 - If the code is valid, the user hears a splash tone to indicate the account code was accepted. The call continues and the code is added to the Call History record.
 - If the code is too long or too short, the caller hears, "Account codes must be X digits. Please re-enter your X-digit account code."

Allow Call to Proceed After Max Retries

The max retry limit is set in **Account Codes Settings**. Once the caller reaches this limit on an outbound call, they hear:

- **On (Allow):** "You have reached the maximum number of attempts. No account code will be assigned to this call." And the call continues without an account code.
- **Off:** "You have reached the maximum number of attempts to enter a valid account code. Goodbye." The call is disconnected.



Call Types

Choose which specific call types require an account code. These optional settings are typically used to ensure that only authorized personnel can place certain types of calls that may incur additional charges.

If one or more call types are enabled, only those call types require an account code. However, if no call types are enabled, all calls require an account code.

- **International.** Require a code to authorize calls to country codes that are different than the caller's. Since the US and Canada have the same country code, calls between the countries are not considered international.
- **Directory assistance.** Require a code to authorize calls to directory assistance (411 and 0).
- **Premium toll.** Require a code to authorize calls to 1-900 numbers.



Validation Setting

To prevent users from placing these call types without a valid code, ensure the "Validate account codes when dialed" setting is enabled; otherwise, the call will go through even if the number dialed doesn't match a code on the account.

Use Cases

Settings	Results
Account Codes is not yet live.	
 Account Codes is enabled for the account. 	Account codes are not prompted for outbound calls.
Account Codes is not enabled for the SIP trunk.	• Account codes are not allowed for inbound calls.
Account codes are optional.	
 Account Codes is enabled for the account and SIP trunk. 	• Account codes are not prompted for calls that originate and terminate on the
 Require account codes for internal calls: OFF Calls requiring account codes: OFF 	 Account codes are prompted for outbound calls to the PSTN.
	 Account codes are allowed for inbound internal calls.

Settings	Results
Account codes are required.	
 Account Codes is enabled for the account and SIP trunk. 	• Account codes are prompted for outbound calls to the PSTN.
 Require account codes for internal calls: OFF Validate account codes when dialed: ON Calls requiring account codes: OFF 	 Account codes are not prompted for calls that originate and terminate on the same account (internal calls). Account codes are allowed for inbound internal calls. A valid account code is required. If the code dialed is invalid, the call will not proceed.
Track internal calls only.	
 Account Codes is enabled for the account and SIP trunk. Validate account acdes when dialed: 	 Account codes are prompted for outbound calls to the PSTN. Account codes are not prompted for
 Validate account codes when dialed. OFF Require account codes for internal calls: 	calls that originate and terminate on the same account (internal calls).
ONCalls requiring account codes: OFF	 The account code dialed can be any number, so long as it is the correct length.
	• Account codes are allowed for inbound calls.
Track all outbound calls.	
Account Codes is enabled for the account and SIP trunk.	• Account codes are prompted for all outbound calls.
 Validate account codes when dialed: OFF 	 Account codes are allowed for inbound calls.
 Require account codes for internal calls: OFF Calls requiring account codes: OFF 	• The account code dialed can be any number, so long as it is the correct length.

Settings	Results
Require authorization for all outbound calls.	
 Account Codes is enabled for the account and SIP trunk. 	• Account codes are prompted for all outbound calls.
 Validate account codes when dialed: ON Calls requiring account codes: OFF 	 Account codes are allowed for inbound calls. A valid account code is required. If the code dialed is invalid, the call will not proceed.
Require authorization for international calls.	
 Account Codes is enabled for the account and SIP trunk. Validate account codes when dialed: ON Calls requiring account codes: International: ON Directory assistance: OFF Premium toll: OFF 	 A valid account code is required to place a call to international locations. If the dialed code is invalid, the call will not proceed. All other calls are <i>not</i> prompted for an account code. Account codes are allowed for inbound calls.
Require authorization for calls to 1-900 numbers.	
 Account Codes is enabled for the account and SIP trunk. Require account codes for internal calls: OFF Validate account codes when dialed: ON 	 Account codes are not prompted for outbound calls to the PSTN. Account codes are not prompted for calls that originate and terminate on the same account (internal calls). A valid account code is required to
 Calls requiring account codes: International: ON Directory assistance: ON 	 place a call to an international location, directory assistance, or a premium toll number. Calls to local and national long-distance
 Premium toll: ON 	numbers are <i>not</i> prompted for an account code.

Account Codes Setup

Account Codes settings are managed at the account level. Once it's enabled on the account and the default settings are configured, you can choose whether to enable Account Codes for all (current) SIP trunks, after which it can be enabled or disabled for individual trunks as needed.

Enable Account Codes on Account

The following steps are only required the very first time Account Codes is enabled on an account:

- 1. Go to the **Account Codes** page.
- 2. Fill out the following fields:
 - Account code length: Specify the number of digits (3–7) that will be required for all codes on this account. This setting cannot be changed later, so choose a length that will meet the account's needs in the long term.
 - **Max retries to validate code:** Specify how many times (1–9) a user can enter an incorrect account code. The industry average is 3–5 attempts.
- **3.** Click **[Save]**. The page is refreshed, and Account Codes can now be configured on the account.

fo create character	account codes, select a code length between 3-7 s. This setting cannot be changed later.
- Account	code length*
4	
– Max retr	ies to validate code*
3	
3	

Account Settings

Once Account Codes is enabled on an account, the account-level settings must be configured before enabling it on SIP trunks. These settings are applied to all users who have account codes enabled, except those with custom settings.

Account Codes work differently depending on the settings. For details about how to configure your settings for different use cases, see Account Codes Settings.



- 1. Go to Account > Account Codes.
- 2. Under Account Codes Settings, switch on Allow account codes. Once these settings are saved (see step 6), Account Codes can be
- **3. Settings.** Choose which settings to enable (optional). To see how each of these settings changes the flow of a call, see Account Codes Settings.
 - **Require account codes for internal calls.** If enabled, users are prompted to dial an account code on every outbound call to other users on the account. If disabled, the user can dial a star code + account code during a call if they want to tag it.
 - Validate account codes when dialed. If enabled, the account code is validated against codes on the account. If disabled, the user can dial any number as an account code, as long as it's the right length. At least one account code must be added before enabling this option.
 - Allow call to proceed after max retries. If account codes are validated and the user enters multiple invalid codes, this setting determines whether the call will proceed as intended or end once the user reaches the max retry limit. This setting can only be customized at the user level if it's enabled at the account level first.
- **4. Calls requiring account codes.** Choose which specific call types require an account code. These settings are typically used to ensure that only authorized personnel are able to place certain types of calls that may incur additional charges.

To prevent users from placing these call types without a valid code, ensure "Validate account codes when dialed" (above) is enabled. Otherwise, the call will go through even if the number dialed doesn't match a code on the account.

- **International.** Require a code to authorize calls to country codes that are different than the caller's. Since the US and Canada have the same country code, they're considered local.
- **Directory assistance.** Require a code to authorize calls to directory assistance (411) and operator assistance (0).
- **Premium toll.** Require a code to authorize calls to 1-900 numbers.



Important

If one or more call types are enabled, only those call types require an account code. However, if no call types are enabled, *all calls* require an account code.

5. Click [Save].

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\$

- 6. Save Account Codes. A pop-up will appear stating that these settings will be applied to all trunks that have account codes enabled, except those with custom settings, and ask if you want to enable this feature for all users who don't have it yet.
 - **Yes:** Account Codes will be enabled for all existing SIP trunks that don't have it yet. If Account Codes had been intentionally disabled for any SIP trunks, it will have to be disabled for them again.
 - No: Account Codes will *not be* enabled for SIP trunks that don't have it yet. Choose this option if you want to enable it manually for individual trunks, rather than enabling it for all of them at once.
- 7. Click [Save] to apply the settings.

Save Acc	ount Codes
These settin codes enabl	gs will be applied to all users who have account ed except those with custom settings
Do you also have it yet?	want to enable account codes for users who don't

Enable Account Codes on SIP Trunk

The default settings configured at the account level apply to all SIP trunks that have Account Codes enabled. When Account Codes is enabled, callers are prompted to dial a code on every outbound call. Whether or not the code must be valid (preconfigured on the account) depends on the account's settings. See How to Use Account Codes for details.

When Account Codes is enabled for the account, it can automatically be enabled for all existing SIP trunks. However, it is *not automatically enabled* for new trunks, so it must be enabled manually for those that need it.

- 1. Go to the **SIP Trunks** page and select the trunk you want to customize.
- 2. Scroll down to Account Codes.
- **3.** Check *Enable account codes* to turn it on. Once saved, callers will be prompted to dial an account code on every outbound call placed from the trunk.



4. Click **[Save]**. Account Codes is now enabled with the account settings.



Disable Account Codes

To disable Account Codes on a SIP trunk, uncheck "Enable account codes". To disable the feature for the account (and all trunks), see Disable Account Codes.

Manage Account Codes

Account Codes allow users to tag calls for billing purposes or authorize calls to international locations, directory assistance, or premium toll numbers. Create as many account codes as needed.

For calls that are prompted for an account code, if the "Validate account codes when dialed" setting is enabled, the user must dial a code that is set up on the account or the call will not proceed. However, if this setting is not enabled, the user can dial any code they want, whether or not it's set up on the account, as long as it's the right number of digits.

At least one code must be added to the account to enable this setting; otherwise, users won't be able to place outbound calls. Unless Account Codes are being used to authorize certain calls, we recommend creating at least one account code, such as 1000, that can be used as a default option.



Account Codes		+ Account Code
Q Search account codes		
Name or Description	Account Code	
Demo Account Code 1	101	/ 1
Demo Account Code 2	102	/ 1
	Items per page: 20 💌 1 - 2 of	f2 < < >>



Important

Account codes cannot start with zero (0).



End User Access

End users don't have access to the list of account codes, so you'll need to provide them manually. Since codes cannot be assigned to specific users or for specific purposes, only give users the code(s) you want them to have.

Add Account Code

- 1. Go to the Account Codes page.
- 2. Click [+ Account Code] in the top right.
- 3. Enter a name or brief description to remember what the code is assigned for.
- 4. Enter the code that will be dialed.
- 5. Click [Create].
- 6. Repeat steps 2–5 for any additional codes.

New Account Code	
- Name or description*	
Account code (4 digits)*	0
	Cancel Create

Edit Account Code

Click the edit icon to update the name or description of an existing account code. To change the number, create a new one and then delete the old one.

Edit Account Code	
Account code*	0
	Cancel Save

Disable Account Codes

When Account Codes is disabled at the account level, it is also disabled for all SIP trunks. To disable Account Codes for an individual SIP trunk, go to the trunk settings and uncheck *"Enable account codes"*. To disable it for the account, follow the steps below:

- 1. Go to the Account Codes page.
- 2. Under Account Codes Settings, toggle off Allow account codes and click [Save].
- **3.** Click **[Save]** to disable Account Codes for all SIP trunks on the account. The account codes are preserved, so they can be used again if the feature is re-enabled later.

Account Codes Settings	
Account code length 4	
 Max retries to validate code* ——— 	
3	
3	
Allow account codes	
Save	

Account code custom user s	s will be disabled for all users on the account. Any ettings will be preserved in case account codes is
e-enabled in 1	he future, unless you choose to delete those
ettings below	Ι.
Delete cu	stom user settings

Local Services

SIP Trunk Local Services enables the SIP trunk to translate local service calls (like 211, 411, etc.) from a 3-digit number to a 10-digit number, so it doesn't have to be configured on the PBX.

Local Services is enabled on a per-trunk basis. Once enabled, the voice server will accept and route 3-digit N11 codes (411, 611, etc.) and 7-digit local dialing numbers from the SIP trunk. SIP trunks will no longer need to translate these numbers into 10-digits before they're sent to the voice server.

- **Emergency Numbers.** Whether or not Local Services is enabled, calls to 911, 933, and 988 will route correctly. Geographic data used for public service functions and 7-digit dialing will be assigned to the SIP trunk based on its primary phone number.
- International Numbers. After enabling Local Dialing, users will need to dial 011 before international numbers, even if the SIP trunk was previously configured to allow international dialing without it.
- **N11 Numbers.** In some cases, Local Services must be enabled for SIP trunk users to dial N11 codes. Refer to Social & Public Services for more detailed information.

Pattern	Description
International Long Distance	011+
National Long Distance	1+
Local Dialing	7- and 10-digit numbers, translated to 1+NPA+dialed number
7-digit Local Dialing	Translated to 1+NPA+dialed number
10-digit Local Dialing	Translated to 1+dialed number

Supported Dialing Patterns



Note

After enabling Local Dialing, users must dial 011 before international numbers, even if the SIP trunk was previously configured to allow international dialing without it.

Enable Local Services

Getting started is a two-part process: Enabling Local Services in the Alianza Admin Portal and ensuring the end user's SIP trunk/PBX is configured to use it.



- 1. In the Admin Portal, search for the account and then go to Account >Trunks.
- 2. Locate an existing trunk and click **Edit**, or click **[+ Add SIP Trunk]** and fill out the information to set up a new one.
- 3. Under **Details**, check **Enable Local Services** and select a primary phone number.

Local Services	1	2005	Unregistered C	^
Details				
Trunk Name	Local Services			
Registration Status	No ×			
Assigned Phone Numbers	1 (205) 476-4466			
911 Callback Number	1 (205) 476-4466 - 138 S 800	W, Orem, UT 84058	-	
Primary Phone Number	1 (205) 476-4466 - 138 S 800	W, Orem, UT 84058	-	
	Used for identifying outbound cal services (not used for caller ID).	ls from this SIP trunk for billing	g and local	
Enable Local Services	 Route N11 calls to local s 	ervices (learn more)		

A primary phone number is not required, but it is recommended. Alianza determines where to route local service calls based on the address associated with this number.

If you're setting up a SIP trunk for the first time and a primary phone number is not available, select *None* and update this field when a phone number becomes available. In the meantime, be sure to provide geographic information to the SIP trunk configuration; otherwise, local services dialing will fail, including Directory Assistance, Call Public Services, and 7-digit dialing.

- **4.** Save the SIP trunk configuration.
- 5. Next, update the end customer's SIP trunking device to allow the new configuration. This process will vary for each device.

Social & Public Services

The following N11 codes provide quick three-digit dialing access to special services in the United States and Canada, based on the caller's address, without the need for an area code.

Code	Description
211	Essential Community Services
	Access to community information and referral services, such as essential needs, crisis, and disaster assistance. Visit http://www.211.org/ to learn about services in your area.
411	Directory Assistance
	Phone service used to look up a published telephone number and/or address listing.
511	Traveler Information (US)
	Local hotline for real-time information regarding traffic and road conditions. Not available in all states.
611	Customer Service
	Dials Customer Service.
711	Telecommunications Relay Service
	TRS uses operators to facilitate phone calls between people with hearing and speech disabilities and other individuals. A TRS call may be initiated by a person with or
	without a disability. Visit http://www.fcc.gov/ to learn more.
811	Utility Location Services (US)
	"Call Before You Dig" routes the caller to their local utility location services. Call a few days before beginning an excavation project to find out the location of underground utilities and reduce the risk of serious damage.
811	Canadian Health Services (CAN)
	Call to speak to a local health care professional about medical advice, mental health, healthy eating, and more.
911	Emergency Services
	Calls to 911 (US or CAN) are sent to the nearest Public Safety Answering Point (PSAP) based on the registered address. The callback number and address are available to the PSAP on each call.
933	Emergency Services Validation
	Calls to 933 are sent to the caller's emergency services provider, who will then connect the call to their automated 911 verification service. The service will play back the dialing phone number and the address associated with it.

Code Description

988 Suicide Prevention Hotline

Contact the National Suicide Prevention Lifeline (US) or Talk Suicide Hotline (CA) to speak with a trained crisis counselor who will listen, offer support, and get them the help they need.



Canadian Support

Not all Canadian rate centers support all N11 service codes. If you are unable to call a particular service, it may not be available in your area.

Trunk Groups

SIP trunk groups are essential for businesses seeking to optimize their voice communication infrastructure. By grouping trunks together, organizations can quickly update the routing settings for all the trunks in just a few clicks, streamlining the administration, configuration, and maintenance of their voice services into one process.

For example, if trunks are grouped by region and one region experiences a large influx of calls, it would take only a few clicks to update the settings for all trunks in the group to fail over to another SIP trunk or even another region (group), eliminating all the time that would have been needed to update each trunk individually.

Shared vs Independent Settings

SIP trunks assigned to a group share call forwarding, overflow routing, and inbound failover routing settings. These group settings take precedence over the configuration set on the individual SIP trunks within the group. However, all other settings continue to be managed at the individual trunk level.

See Manage SIP Trunks for details.

Shared Group Settings	Independent SIP Trunk Settings
Forwarding settings	IP authentication
Overflow routing	Assigned phone numbers
Overflow trunks	• 911 callback number
Forward on overflow settings	Primary phone number
Inbound failover routing	Local Services
Failover trunks	Dialing digits
Forward on failure settings	Concurrent calls allowed
	Assigned extensions
	Calling plans (see Shared Calling Plans)

Add or Manage SIP Trunk Group

Log in to the Voice Portal and go to **Account > SIP Trunks**. From here, select an existing trunk to view or modify the settings, or click **[+ Add SIP Trunk]** in the top right to create one, then fill out or modify the information below.



Note SIP tru

SIP trunks must already be created on the account before they can be added to a group. See Manage SIP Trunks for details.

Group Name

Enter a name for the group that is less than 25 characters.



Forward Settings

Check this box to forward *all* inbound calls intended for the SIP trunks in this group to a specific destination. Forwarding destinations include:

- User (select a user)
- Auto-Attendant (select an auto-attendant)
- Voicemail (select a voicemail box)
- Telephone Number (enter an on- or off-net phone number)
- Busy Signal



Overflow Routing

When a SIP trunk in the group reaches capacity, meaning it has as many calls as allowed by the concurrent call setting on the individual SIP trunk, calls are rerouted to the first available trunk in this list. If all trunks reach capacity, calls are forwarded to the overflow destination below.

• Add Overflow SIP Trunk. Select a SIP trunk that inbound calls will fail over to once a trunk in this group reaches capacity. Add as many trunks as needed in priority order. To **reorganize** the SIP trunks, click — and drag it to the new location.



- **Forward on overflow.** If all SIP trunks in the group reach the concurrent call limit, calls will be forwarded to the following destination:
 - User (select a user)
 - Auto-Attendant (select an auto-attendant)
 - Voicemail (select a voicemail box)
 - Telephone Number (enter an on- or off-net phone number)
 - Busy Signal

ty,

Inbound Failover Routing

When a SIP trunk in the group fails, meaning it cannot register to the platform, calls are rerouted to the first available trunk in this list. If all trunks fail, calls are forwarded to the failure destination below.

Note

Calls being routed to SIP trunks that are not in this group will *not* follow that SIP trunk's fail over routing configuration.

- Add Failover SIP Trunk. Select a SIP trunk that inbound calls will fail over to. Add as many trunks as needed in priority order. To reorganize the SIP trunks, click — and drag it to the new location.
- Forward on failure. Select what happens to inbound calls if all trunks fail:
 - User (select a user)
 - Auto-Attendant (select an auto-attendant)
 - Voicemail (select a voicemail box)
 - Telephone Number (enter an on- or off-net phone number)
 - Busy Signal

Vhen a SIP tru vailable trunk he failure des	unk in the group fails, calls are rerouted to the first c in this list. If all trunks fail, calls are forwarded to stination below.
	Failover SIP Trunk
=	Park Lane Building C 🔹
	+ Failover SIP Trunk
– Forward on fa	ailure
User	v
User	

SIP Trunks

- 1. In the SIP Trunks section, do one of the following:
 - To **add** a SIP trunk to the group, click **[+ SIP Trunk]** and select a trunk.
 - To **remove** a SIP trunk from the group, click **Trash** on the right.
 - To **reorganize** the SIP trunks, click and drag it to the new location. Functionally, the order of the trunks doesn't matter, but you can organize them as you'd like.
- 2. When you're done, click **[Save]**. The changes will be applied immediately.

SIP Trunk	S			
-	Park Lane Building A	*	ī	
-	Park Lane Building B	•	Î	
=	Park Lane Building C	•	Î	
+ SIP Trunk				
Cancel			Save	

"cymbus

Analytics

The inventory dashboard, viewable only to Account Managers and Super Admin users, displays a read-only view of the products, licenses, and add-ons in an account.

Each product is organized by name, the account status (Active, Suspended, or Disabled), the date the status was last updated, and any packages and/or add-ons that are included. Listed below each license or add-on is the number of associated users or lines.

Inventory			
Developed			
Product			
Sip Trunk			
Active · Apr 30th, 2024			
	-		
Licenses			
Call Path			
31			

Call History

Call History holds the records of all calls made and received on the account. Calls are listed in chronological order with the most recent call at the top. Call data is organized into columns that show the date and time of the call, where the call originated (From), where the call terminated (To), the call flag for inbound calls, and the account code used.

At the top right of the page, the current record list can be emailed as a .csv file. See Call History Report for details.





Note

The FCC requires that all carriers keep call detail records for a period of 18 months. While some states defer to the FCC, other states have retention rates that exceed the federal requirement. To be safe, we store all call records for a minimum of 3 years.

Apply Call Filters

Apply call filters to locate specific call records. Once your parameters are set, the matching call records are displayed below.

- **Search:** Search for a specific phone number or extension. Results automatically populate after the first three digits are entered.
- **Date Range:** Modify the date and time ranges or delete them entirely to gather the data you need. By default, the last 30 days of call records are displayed.



- **Start/End Time:** Choose the time range, between 12:00 AM and 11:59 PM, based on the account's time zone settings.
- **Filters:** Filter the call records by Direction, Number, Call Flags, and/or Call Category. These filters allow you to identify patterns in the calls, such as how many calls are being sent to voicemail, what calls are being missed, etc.
- **Reset:** When you're finished with your search, click **Area Reset Filters** to remove any parameters and display all calls.



Select Call Options

To view a call's details, click the menu on the right. The option to block the number is also available for inbound calls.

PM EDT	SAN JOSE-SUNNYVALE-SANTA	DALLAS-FORT WORTH-	Missed	None	
) seconds	CLARA, CA	ARLINGTON, TX			C
April 11, 2025 at 1:00:02 PM EDT 2 minutes	1 (778) 555-1234 VANCOUVER, BC	1 (515) 555-0987 AMES, IA	None	Block No Detai	Number Is
April 11, 2025 at 12:56:30 PM EDT 2 minutes	1 (778) 555-7654 VANCOUVER, BC	1 (515) 555-6543 AMES, IA	None	None	:

Block Number

If necessary, you can block an inbound caller from making additional calls to the account.

- 1. Click [Block Number].
- 2. Select whether the number should be blocked on the user or the account.
- 3. Click [Confirm] to add the inbound phone number to the blocked caller list.

Details

Call details include the date, time, and length of the call, as well as the cost, origination, and termination information.

Field	Description
Time and Length	When the call started, connected, and ended, what duration of the call was billed, and the actual length of the call.
Cost	The cost of the call, if the call was within the calling plan, and the rate per minute from the plan.
Origination	The number and location of the originating call, and if the call came from on or off the network.
Termination	The number and location of the user that received the call, if they were on or off the network, and how the call was flagged.

Call History Report

Once you've applied the appropriate filters to locate the data you need, you can export the data into a .csv file for offline use.

1. In the account, go to Call History.



- 2. Filter the table as needed.
- 3. At the top right of the page, click [™] Email CSV [Email CSV]. A banner is displayed on the page to indicate that your report is being generated.
- **4.** When the file is ready, an email with a link to download the report is sent to the email address in your settings. The link **expires in 7 days** and can only be used once.

Here's an example of what it looks like:



Report Fields

This list includes all the fields available in the Call History report which can be emailed (exported) from the portal as a .csv file. Each line in the file represents a call.

See Call History for information about how to access these records in the Voice Portal.

#	Field Name	Туре	Description
А	AccountBillableAbbrRCName	string	Abbreviated name of the rate center of the account billable phone number.
В	AccountBillableCityName	string	Name of the city of the address associated with the phone number.
С	AccountBillableLocation	string	MSA (metropolitan statistical area) of the address associated with the number.
D	AccountBillableNumber	string	A phone number on the account.
E	AccountBillableState	string	State of the address associated with the rate center.
F	AccountCode	string	The account code tagged on this call.



#	Field Name	Туре	Description
G	AccountNumber	string	The client-assigned account number of the account to which the call is billed.
Н	Acctld	string	Unique ID of account to which the call is billed.
Ι	ActualCallLengthSeconds	numeric	The length, in seconds, of the call from connect to end.
J	BillCallLengthSeconds	bigint	The length, in seconds, of the call from connect to end rounded according to applied calling plan product.
К	BillingCode	string	Unique code to identify the partition responsible for billing.
L	CallFlagType	string	 Indicates if the call was answered on a device, and if not answered on a device, how did it terminate. Answered Busy Forwarded Missed Voicemail
Μ	CallPickupFromId	string	Indicates the object that picked up the call.
Ν	CallPickupById	string	Indicates the object from which the call was picked.
0	CallType	string	Inbound or outbound.
Ρ	CallingPlanProductId	string	Unique ID of the calling plan product with which the call was rated.
Q	CallingPlanProductName	string	Name of the calling plan with which the call was rated.
R	ConnectTime	date/time	Date and time the call was connected. Billing starts at this time.

#	Field Name	Туре	Description
S	Cost	double	Rated value of call, according to calling plan product.
Т	DialedNumber	string	The digits originally dialed to start the call.
U	DisconnectType	string	Indication of which party disconnected the call; "HangUp" indicates the origin, "HangUpOther" indicates termination.
V	EndTime	date/time	Date and time the call ended.
W	ForwardingNumber	string	Phone number the call was forwarded to.
Х	ForwardingNumberAbbrRCName	string	Abbreviated name of the rate center of the phone number the call was forwarded to.
Y	ForwardingNumberCityName	string	Associated rate center city name of the phone number the call was forwarded to.
Ζ	ForwardingNumberLocation	string	MSA (metropolitan statistical area) of the rate center of the number.
AA	ForwardingNumberState	string	State of the associated rate center of the phone number the call was forwarded to.
AB	Id	string	Unique string of characters assigned to each call within the Alianza system.
AC	IdentityAttestLevel	string	STIR/SHAKEN field. A = fully attested or trusted source, B = partially attested, or C = not attested (potential spam).
AD	IdentityOriginationId	string	STIR/SHAKEN field. A unique identifier used to identify the source of the call.
AE	IdentitySignOrganization	string	STIR/SHAKEN field. The code for the carrier that performed the signing.
AF	IdentitySignSPCode	string	STIR/SHAKEN field. The code assigned to the service provider that signed the call.
AG	InPlan	boolean	True/False indicator of whether the call was considered "in plan."

#	Field Name	Туре	Description
AH	LegType	string	Indication of direction of call:
			Origination
			Forward
AI	MeanOpinionScoreAverage	numeric	Average MOS for the call.
AJ	MeanOpinionScores	list <string></string>	List of MOS scores associated with each SIP call leg.
AK	MediaServerType	string	Indication of what media service was used by call, if any.
AL	MetroServiceArea		Reserved for future use
AM	OrigAbbrRCName	string	Abbreviated name of the rate center of the phone number that made the call.
AN	OrigCallCategory	string	Type of call that was made.
AO	OrigCarrier	string	The name of the carrier, if applicable.
AP	OrigCityName	string	City for the associated rate center for the phone number that made the call.
AQ	OrigLocation	string	MSA (metropolitan statistical area) of the rate center for the phone number that made the call.
AR	OrigNumber	numeric	Phone number that made the call.
AS	OrigState	string	State of the associated rate center for the phone number that made the call.
AT	PartitionId	string	Unique ID of client partition to which the associated account belongs.
AU	RateLocalFromNumber		
AV	RatePerMinute	double	Per-minute rate for call, according to the calling plan assigned to the user making the call.

#	Field Name	Туре	Description
AW	RateType		Indication of why the call was rated the way that it was.
			• Local
			OnPlanMinutes
			OnPlanRated
			OffPlanRated
			• Free
			• TollFree
			• 411
			• Operator
			Local requires the calling plan to be set up with Unlimited Local, and the calls are rated at \$0.
AX	ReferenceId	string	Unique ID of the acting or responsible party on the associated account to which the call was billed.
AY	ReferenceName	string	Name of acting or responsible party.
AZ	ReferenceType	string	Type of object to which the call is billed.
			SIP_TRUNK
			• END_USER
			• ACCOUNT
			BUSINESS_LINE
BA	SessionId	string	Internal softswitch session ID.
BB	SipCallIds	set <string></string>	List of SIP callIds associated with the call.
BC	StartTime	date/time	Date and time the call started ringing.
BD	TermAbbrRCName	string	Abbreviated name of the rate center of the phone number that received the call.
BE	TermCallCategory	string	Type of call that was made.

#	Field Name	Туре	Description
BF	TermCarrier	string	The name of the carrier, if applicable.
BG	TermCityName	string	Associated rate center city name of the phone number that received the call.
BH	TermLocation	string	MSA (metropolitan statistical area) of the rate center of the phone number that received the call.
BI	TermNumber	string	Phone number that received the call.
BJ	TermState	string	State of the associated rate center of the phone number that received the call.
ΒK	OrigCnam	string	The originating caller's name.
BL	TermCnam	string	The terminating caller's name.
BM	ForwardingCnam	string	The forwarding caller's name.
BN	VerStat	string	STIR/SHAKEN field. String from the <i>verificationResponse</i> containing:
			• TN-Validation-Passed. The number passed the validation.
			• TN-Validation-Failed . The number failed the validation.
			• No-TN-Validation. No number validation was performed.

#	Field Name	Туре	Description
BO	VerStatReason	string	STIR/SHAKEN field. String of text from the <i>verificationResponse</i> message used in case of failed verification.
			BAD_IDENTITY_INFO
			CLAIM_TO_SIP_MISMATCH
			INVALID_IDENTITY_HEADER
			MALFORMED_IDENTITY_HEADER
			STALE_DATE
			UNSUPPORTED_CREDENTIAL
			UNSUPPORTED_PASSPORT_FORMAT



Account Settings

Emergency Call Notifications

In recent years, the FCC has passed Kari's Law and RAY BAUM's Act to help expedite response to emergency services to callers and improve outcomes. As part of Kari's Law, all multi-line telephone systems (MLTS) must be preconfigured to send a notification to an on-site location (like a front desk or security office) when a 911 call is made.

For Emergency Call Notifications to work, each phone number on the account must be successfully configured with a valid E911 record, which includes the physical address where the device is located and any other information necessary to precisely identify the caller's location. It is the end user's responsibility to keep this information up to date, but they may need a reminder.

Emergency Notifications	
Emergency Call Notifications	Testing Notifications
Required for all multi-line telephone systems (MLTS) by Karls Law C Atternative solution in use There the email addresses and/or SMS numbers for users at a central location (from desk, security, administrators, etc.) who will be alerted when a 911 call is placed. Email office@telecomdemo.com off	There are two ways to make sure notifications can be received: • Ask the end user to dial 933 • Click the button below A test notification will be sent to the email address(es) and SMS number(s). Check the log below for details. Test Notifications
SMS 1 (385) 555-0459 🔇 1 (801) 555-3995 🔇	
Reset Save	



Alternative Solution

Emergency Call Notifications are *required* for any and all MLTS manufactured, imported, sold, leased, or installed after **February 16, 2020**. If your account already meets this requirement with an on-premises solution, select *Alternative solution in use*.

Configure Notifications

It is important that you choose to notify a central location where someone will see or hear the notification, such as a managed distribution list of on-site personnel (front desk, security office, administrators, etc.), rather than an individual who may or may not be at the location 100% of the time. While there isn't a limit on how many contacts can be entered here, make sure the number is reasonable for your organization.



- 1. Go to **Settings > Emergency Notifications**.
- 2. Identify the email address(es) and SMS-capable phone number(s) that will be notified when an emergency call is placed from a number on the account.
- 3. Enter those email addresses and phone numbers in the portal and click [Save].
- 4. Click **[Test Notifications]** to send a test to make sure it's working.
 - Alternatively, you can ask the end user to dial 933 to verify their emergency call record with their E911 provider. The call will be connected to an automated 911 verification service, which will play back the dialing phone number and its associated address and send a test notification to the ENS recipients.
- 5. Confirm with the recipients they have received the test notifications and they understand what the notification is for.

Now, when someone on your account dials 9-1-1, the emergency call is processed and a notification is sent to the recipients configured in the portal, so they are made aware of the situation and can assist emergency responders upon arrival.

Emergency Notification

The Emergency Notification Log contains a complete history of all test and emergency call notifications sent for this account. The notification type is identified on the left, followed by the date, time, and sent status:

- Successful: The notification was successfully sent to all parties.
- **Partially Failed:** The notification was sent to some but not all parties.
- Failed: The notification was not sent.

1	Emergency Notification Log			
	Notification Type	Date	Sent Status	
	933 Test	12/14/2022 - 08:59 AM	Successful	~
	933 Test	08/17/2022 - 04:11 AM	Partially Failed	~

Expand an entry for details.

Notification Type 个	Date	S	nt Status	
Test Notifcation	02/20/2020 - 12:22 PN	1 Pa	irtially Failed	^
Test Notification				
Email	Status	SMS	S	tatus
jBrady@bado.com	📀 Sent	1 (801) 400-2324	c	Sent
Email2@test.com	8 Failed	1 (801) 400-9284	6	Failed
Emergency Call	02/20/2020 - 12:22 PM	1 St	iccessful	~
Test Notification	02/20/2020 - 12:22 PM	A Fa	iled	~



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