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Legacy SIP Trunking

User 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 Address or Username*

Forgot Password?

Password*

☐ Remember Me

Login

✏️ **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 Password

New Password*

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✓ Includes a minimum of 8 characters

✓ Includes at least one number

✓ Includes at least one special character

Save Password

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.
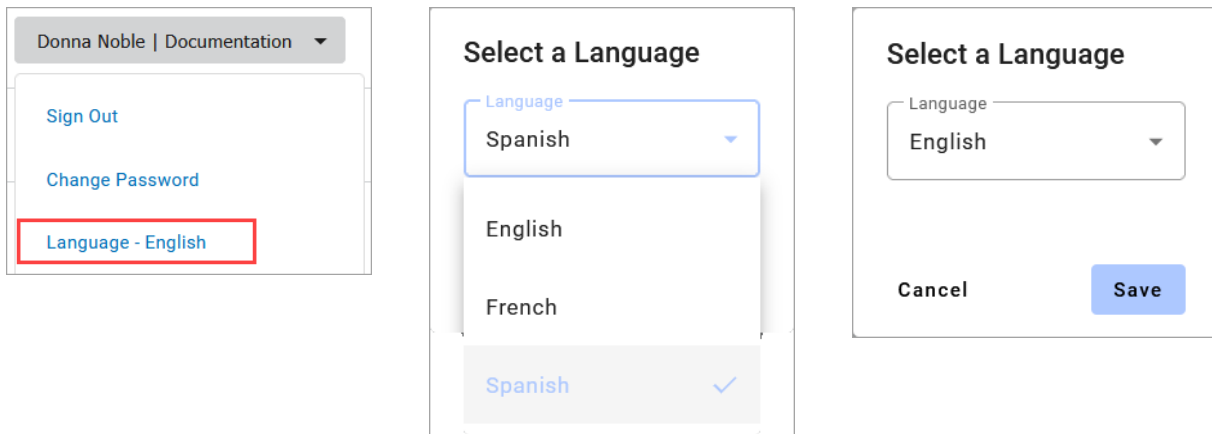
Change Default Language

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 Voice Portal and go to **Account > SIP Trunk**. From here, select a trunk to manage its settings.

Details

Field	Description
Trunk Name	The name of the given SIP trunk.

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Field	Description
Language	Select a language for the trunk, which will be used for audio prompts: • English • French (Canadian) • Spanish
Registration Status	Indicates if the trunk has an active SIP registration with Alianza's platform.
Assigned Phone Numbers	How many and which phone numbers are directed to this SIP trunk.
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.

Forward All

Forward All

Forward All   ☐   All inbound calls will be forwarded (ignores Failover Routing)

Field	Description
Forward All Calls	Check this box to forward all inbound calls intended for this SIP trunk to a specific destination, such as a telephone number, designated below.

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.

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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 when this trunk has reached the concurrent call limit.
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 route calls to the first available SIP Trunk, starting at the top of this list.

Forward on Failure   Busy Signal ▾

If all SIP Trunks fail, calls will be forwarded to this destination.

Field	Description
[+ Add Failover SIP Trunks]	Creates a new menu to select which SIP trunk calls will failover to.
Forward on Failure	If all SIP trunks fail, calls will be forwarded to the configured destination: • User • IVR • Voicemail • Telephone Number • Busy Signal

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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.

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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.

Supported Dialing Patterns

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

> **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.

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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.

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.

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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.

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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.

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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.

Call History   ✉ **Email CSV**

🔍 Search phone numbers

| Date Range | Start time | End time | Filters | ↻ |
| 1/26/2025 − 2/25/2025 📅 | 01:00 AM 🕐 | 11:59 PM 🕐 | | |

Date	From	To	Call Flags	Account Code	
February 24, 2025 at 6:33:46 PM EST 10 seconds	1 (718) 555-0000 NEW YORK-NEWARK-JERSEY CITY, NY-NJ-PA	1 (718) 555-5555 NEW YORK-NEWARK-JERSEY CITY, NY-NJ-PA	Answered	None	⋮
February 24, 2025 at 5:26:50 PM EST 17 seconds	1 (718) 555-1234 NEW YORK-NEWARK-JERSEY CITY, NY-NJ-PA	1 (718) 555-5555 NEW YORK-NEWARK-JERSEY CITY, NY-NJ-PA	Answered	None	⋮

> ✏️ **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 ↻ **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.

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.

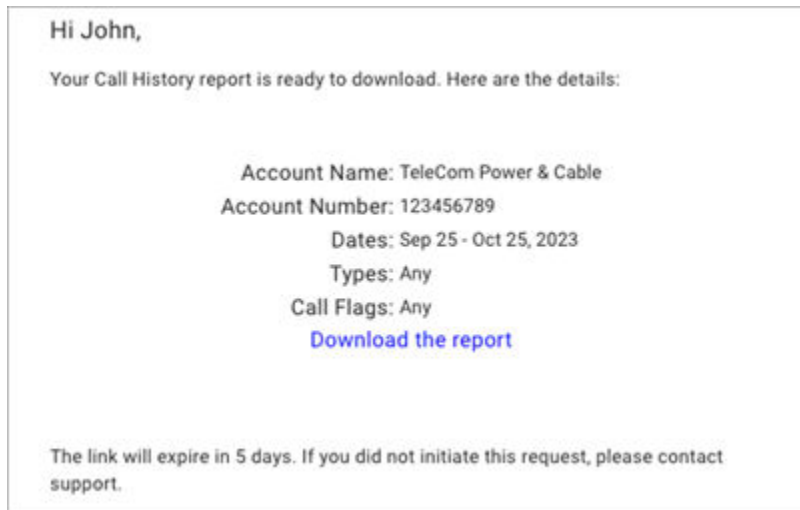| Field	Description
Time and Length	When the call started, connected, and ended, what duration of the call was billed, and the actual length of the call.
Cost	The cost of the call, if the call was within the calling plan, and the rate per minute from the plan.
Origination	The number and location of the 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**.

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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:

   Hi John,

   Your Call History report is ready to download. Here are the details:

   Account Name: TeleCom Power & Cable
   Account Number: 123456789
   Dates: Sep 25 - Oct 25, 2023
   Types: Any
   Call Flags: Any
   Download the report

   The link will expire in 5 days. If you did not initiate this request, please contact support.

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	Type	Description
A	AccountBillableAbbrRCName	string	Abbreviated name of the rate center of the account billable phone number.
B	AccountBillableCityName	string	Name of the city of the address associated with the phone number.
C	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	Type	Description
G	AccountNumber	string	The client-assigned account number of the account to which the call is billed.
H	AcctId	string	Unique ID of account to which the call is billed.
I	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.
K	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
M	CallPickupFromId	string	Indicates the object that picked up the call.
N	CallPickupById	string	Indicates the object from which the call was picked.
O	CallType	string	Inbound or outbound.
P	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.

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#	Field Name	Type	Description
S	Cost	double	Rated value of call, according to calling plan product.
T	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.
X	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.
Z	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."

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#	Field Name	Type	Description
AH	LegType	string	Indication of direction of call: • Origination • Termination • Forward
AI	MeanOpinionScoreAverage	numeric	Average MOS for the call.
AJ	MeanOpinionScores	list<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.

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#	Field Name	Type	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>	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	Type	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.
BK	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 *verificationResponse* 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.

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#	Field Name	Type	Description
BO	VerStatReason	string	STIR/SHAKEN field. String of text from the *verificationResponse* 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.

> ⊘ **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.

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.

Emergency Notification Log

Notification Type ↑	Date	Sent Status	
Test Notifcation	02/20/2020 - 12:22 PM	Partially Failed	∧

Test Notification

Email	Status	SMS	Status
jBrady@bado.com	✓ Sent	1 (801) 400-2324	✓ Sent
Email2@test.com	✗ Failed	1 (801) 400-9284	✗ Failed

Emergency Call	02/20/2020 - 12:22 PM	Successful	∨
Test Notification	02/20/2020 - 12:22 PM	Failed	∨

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