cymbus

Next-Gen SIP Trunking

User Guide

November 2024

Table of Contents

Welcome to SIP Trunking

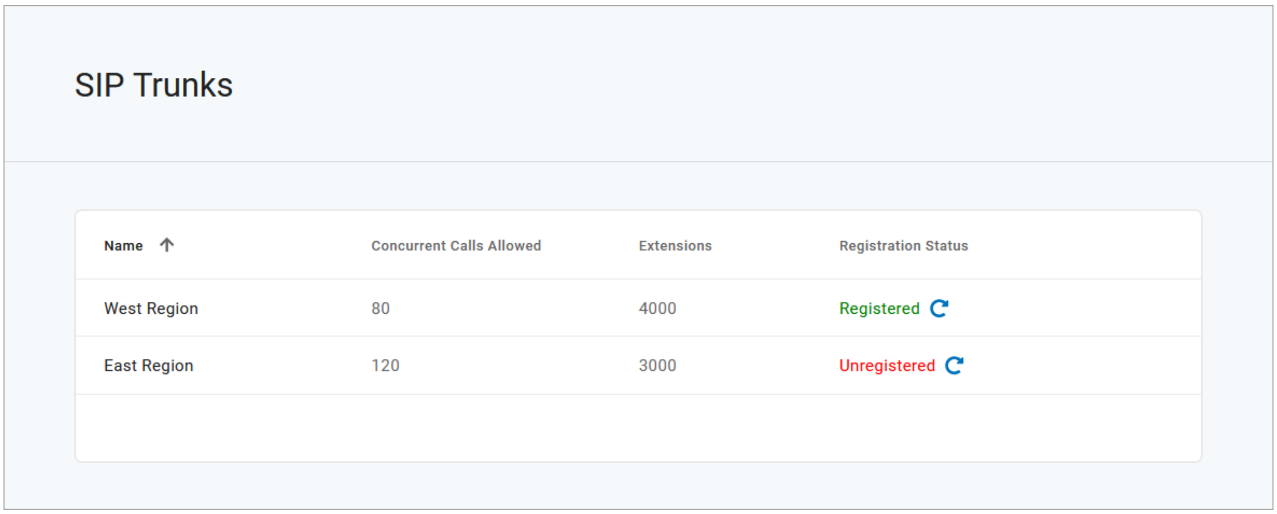
SIP, or Session Initiation Protocol, has become the standard voice connection method in the VoIP industry. A SIP trunk is a connection that carries calls for multiple endpoints, as opposed to most other SIP devices which carry calls for only a single endpoint. Trunks are usually connected to an on-premises 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.

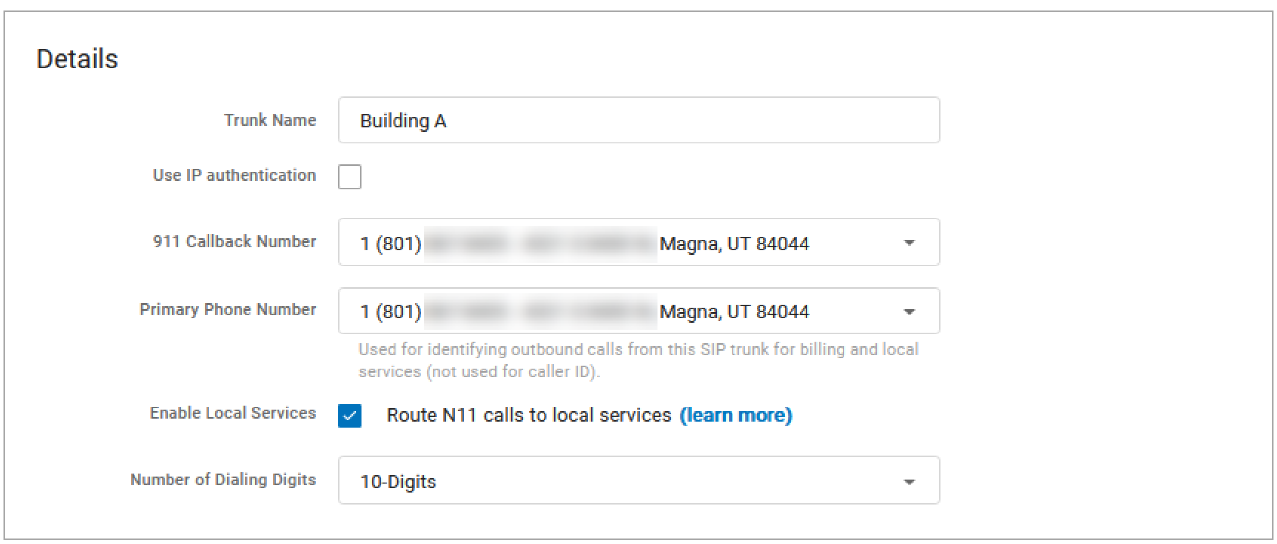
Manage SIP Trunks

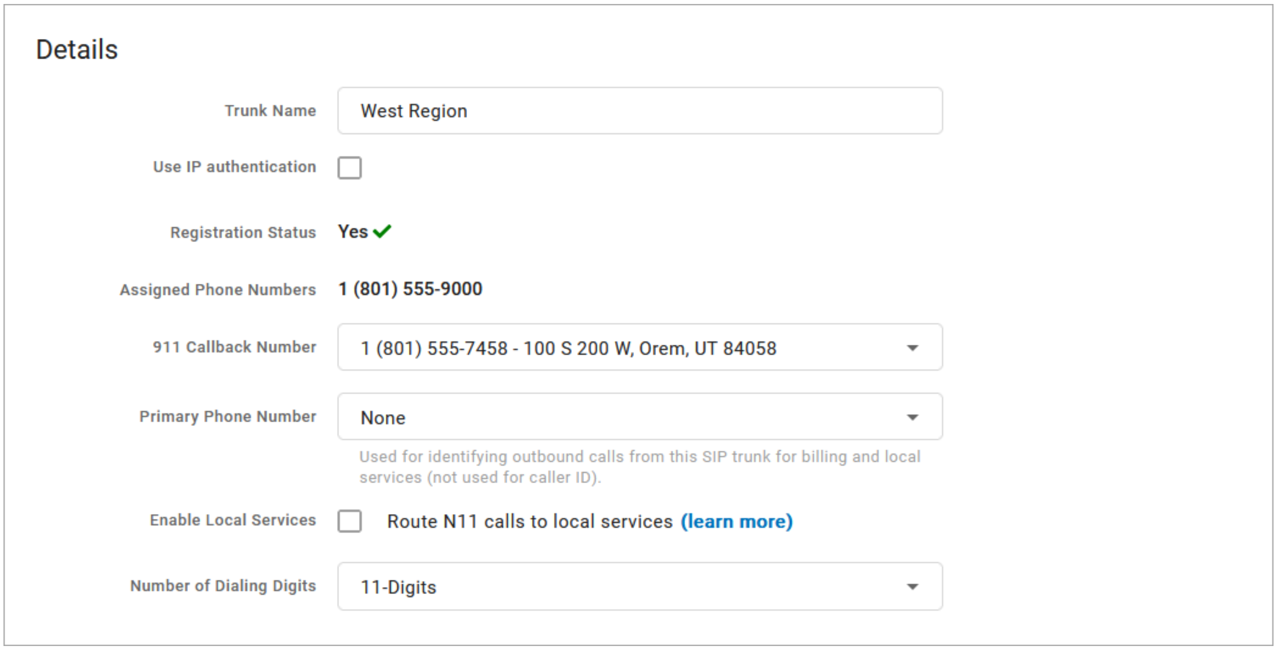
To manage your SIP trunks, log in to the Admin Portal and go to **Account > SIP Trunk**. From here, select an existing trunk to view or modify the settings.

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



Details

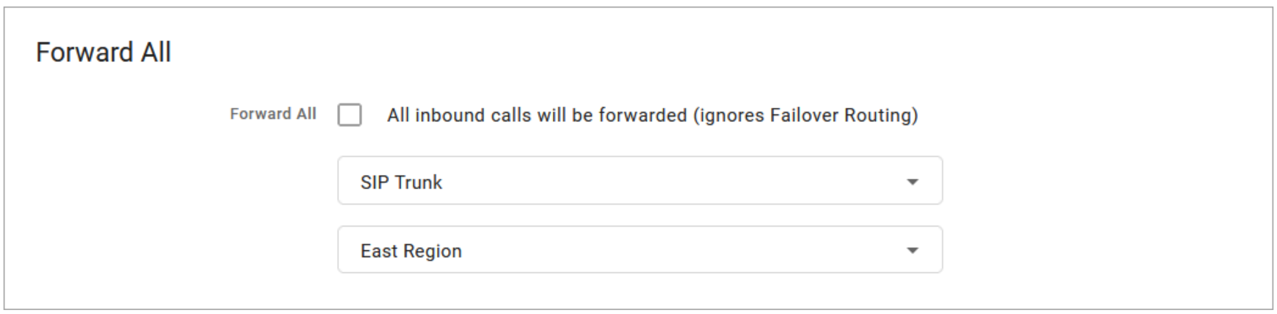




Field	Description
Trunk Name	The name of the given SIP trunk.
Use IP authentication	Identifies whether this SIP trunk uses IP authentication. If unchecked, the trunk will authenticate via user-based SIP credentials, which will be provided in the *Authentication Credentials* section below (once the trunk is saved).
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.

Field	Description
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

Forward All

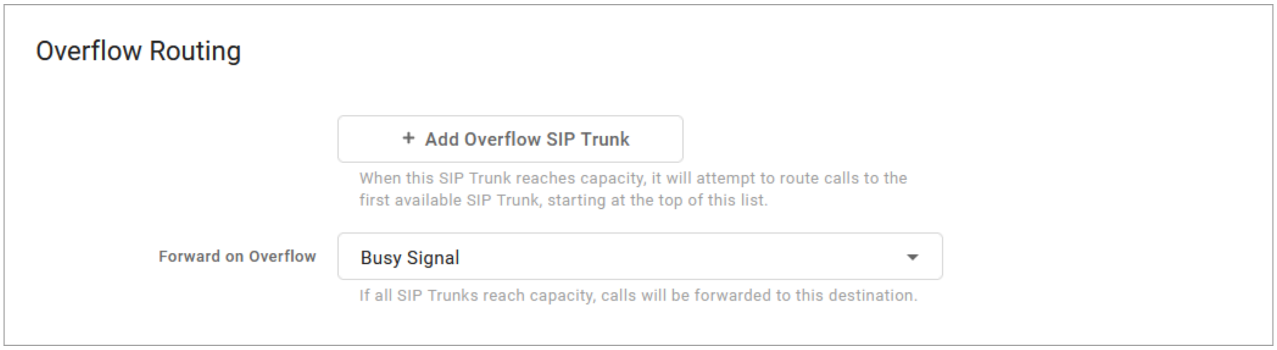


Field	Description
Forward All Calls	Check this box to forward all inbound calls intended for this SIP trunk to a specific destination: • SIP Trunk • User • Auto-Attendant • Voicemail • Telephone Number • Busy Signal

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.

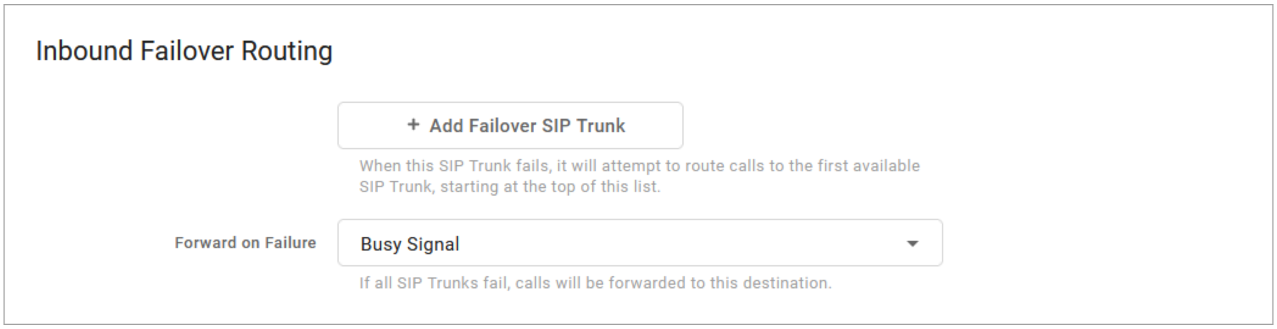


Field	Description
[+ Add Overflow SIP Trunks]	Creates a new menu to select which SIP trunk calls will failover to.
Forward on Overflow	If all SIP trunks reach the concurrent call limit, calls will be forwarded to the configured destination: • User • IVR • Voicemail • Telephone Number • Busy Signal

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.



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

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

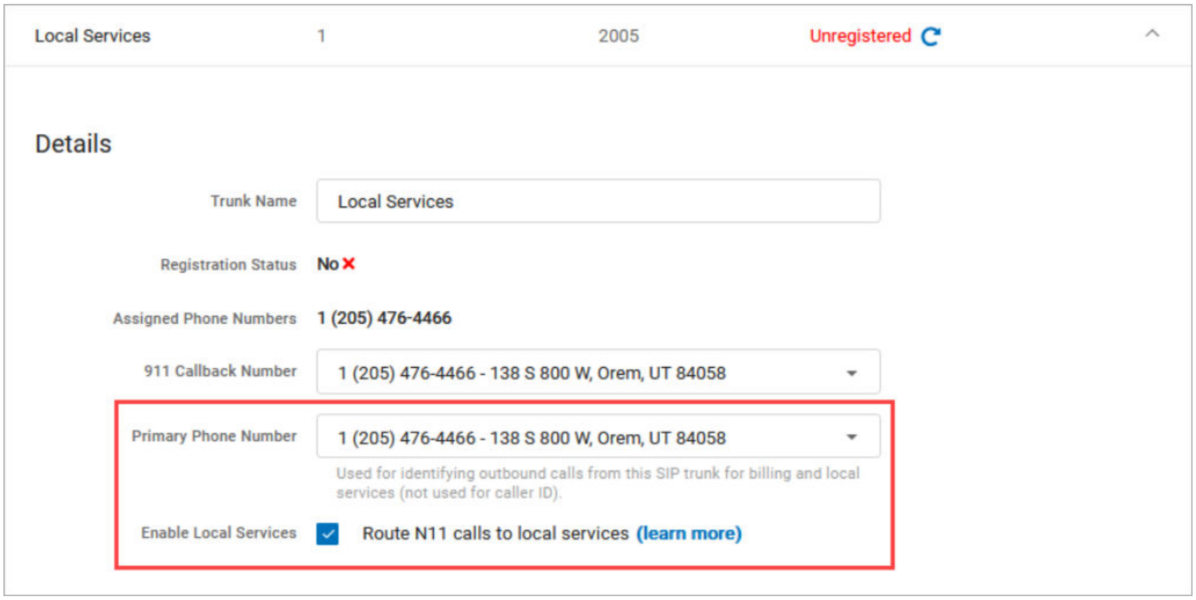
Pattern	Description
International Long Distance	011+
National Long Distance	1+
Local Dialing	7- and 10-digit numbers, translated to 1+NPA+dialed number
7-digit Local Dialing	Translated to 1+NPA+dialed number
10-digit Local Dialing	Translated to 1+dialed number

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.

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 numbers provide quick access to special services based on the caller's location, according to caller ID, 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) will be sent to the nearest Public Safety Answering Point (PSAP) based on the registered address. Both 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 When a user dials 988, they will be connected to 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.

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

Add or Manage SIP Trunk Group

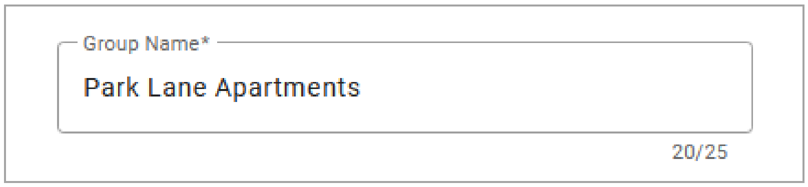
Log in to the Voice Portal and go to **Account > SIP Trunks**. From here, select an existing trunk to view or modify the settings, or click **[+ Add SIP Trunk]** in the top right to create one, then fill out or modify the information below.

Note

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

Group Name

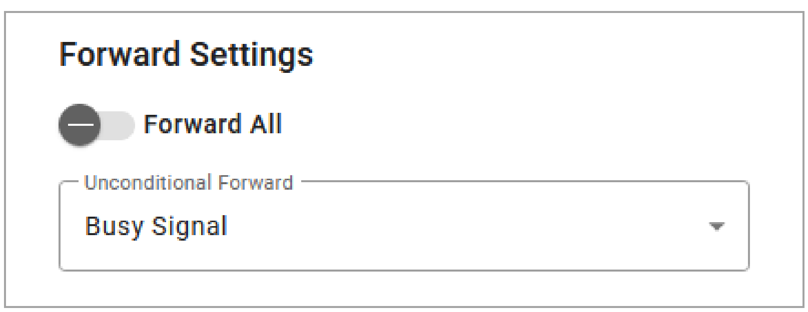
Enter a name for the group that is less than 25 characters.



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:

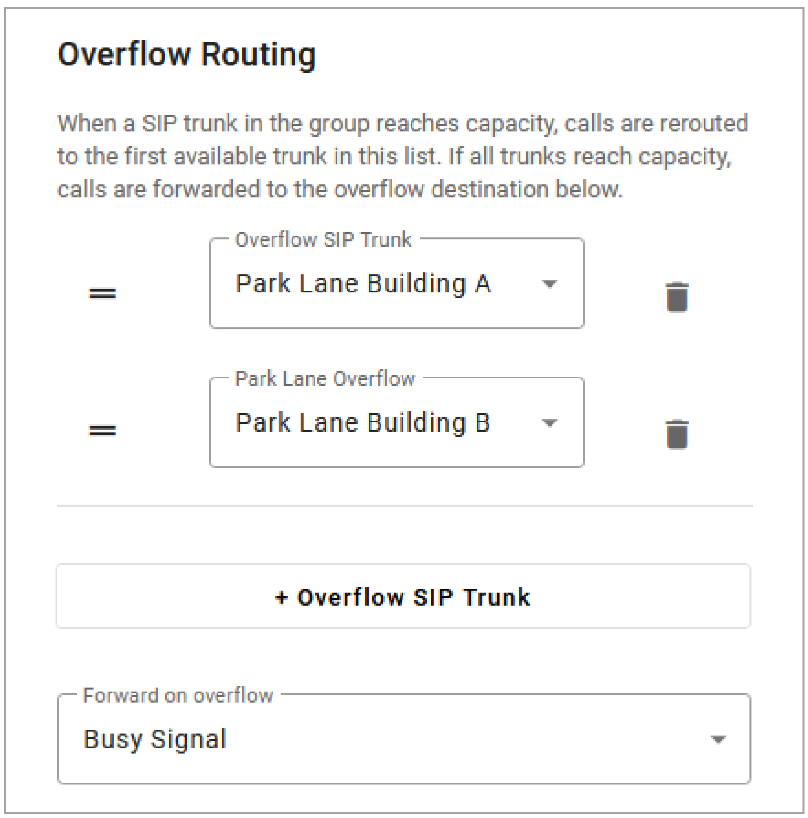
- User (select a user)
- Auto-Attendant (select an auto-attendant)
- Voicemail (select a voicemail box)
- Telephone Number (enter an on- or off-net phone number)
- Busy Signal



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

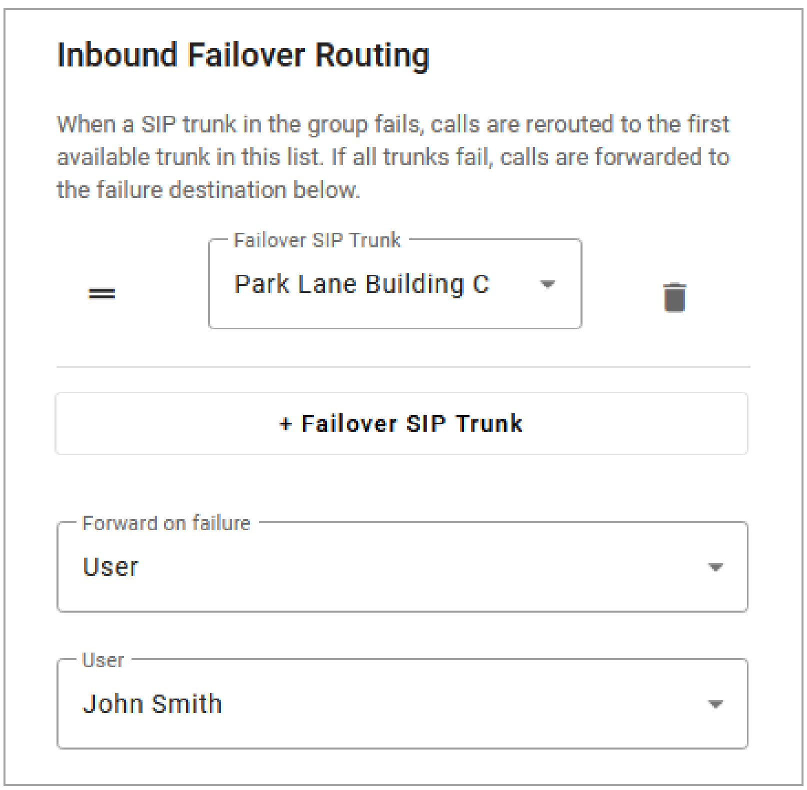


Inbound Failover Routing

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

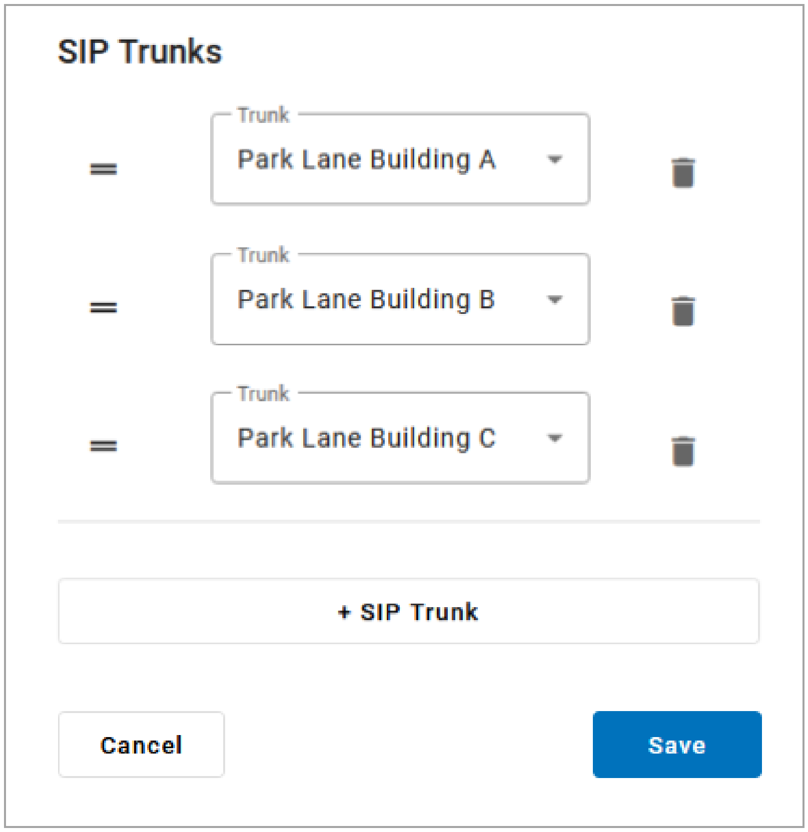


SIP Trunks

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.

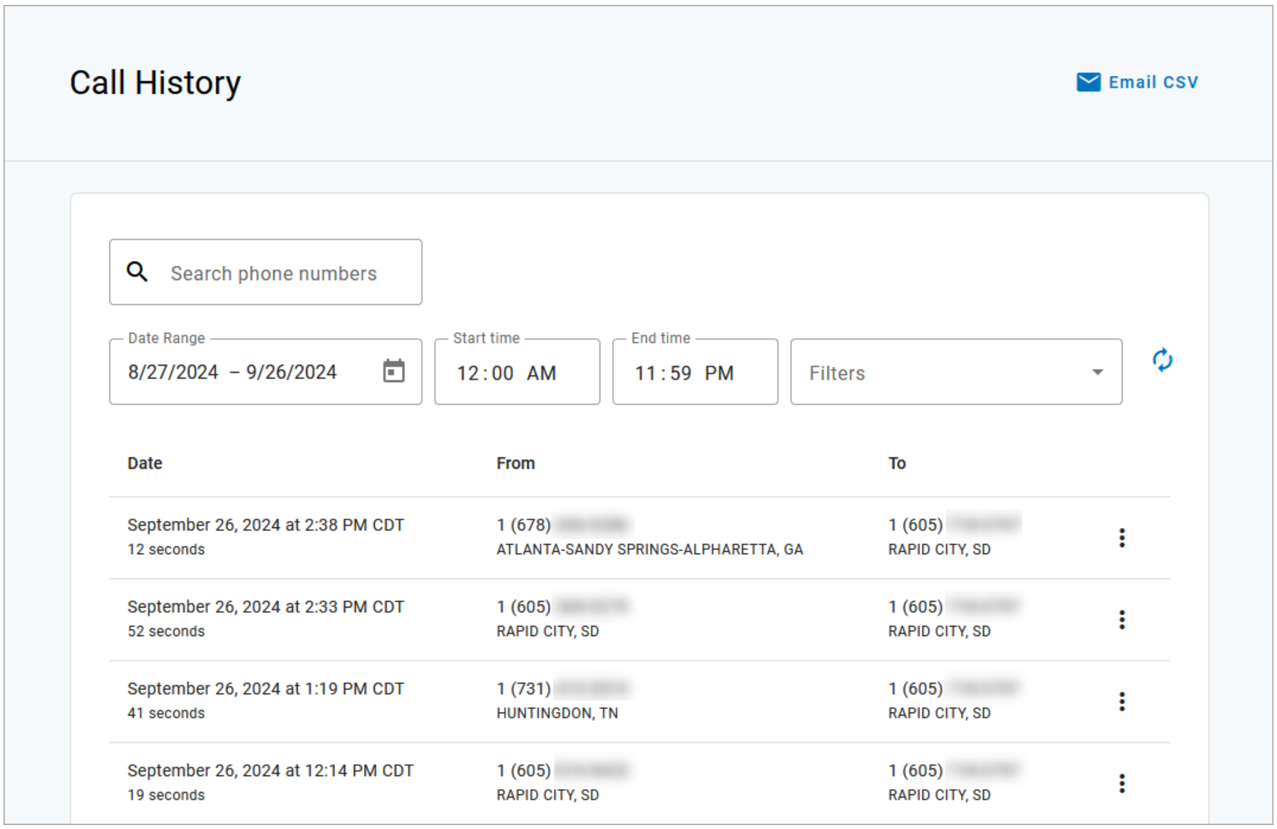
2. When you're done, click **[Save]**. The changes will be applied immediately.



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), and where the call terminated (To).

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.

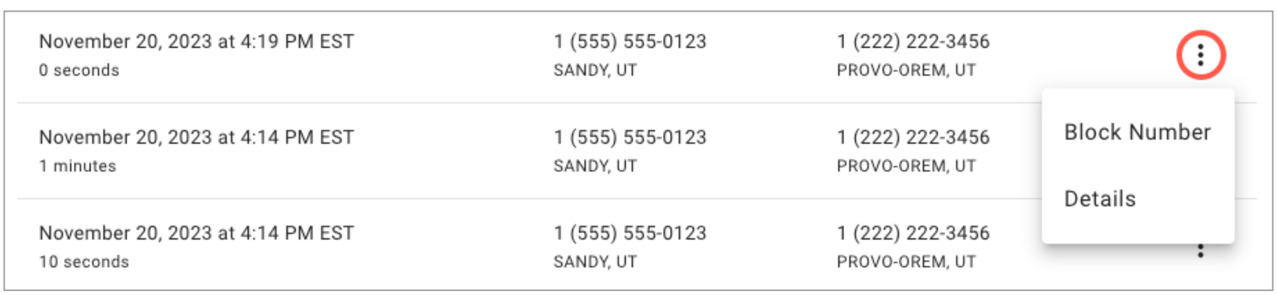
Apply Call Filters

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, and/or Call Flags. 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 either the user or 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, dialed, and termination information.

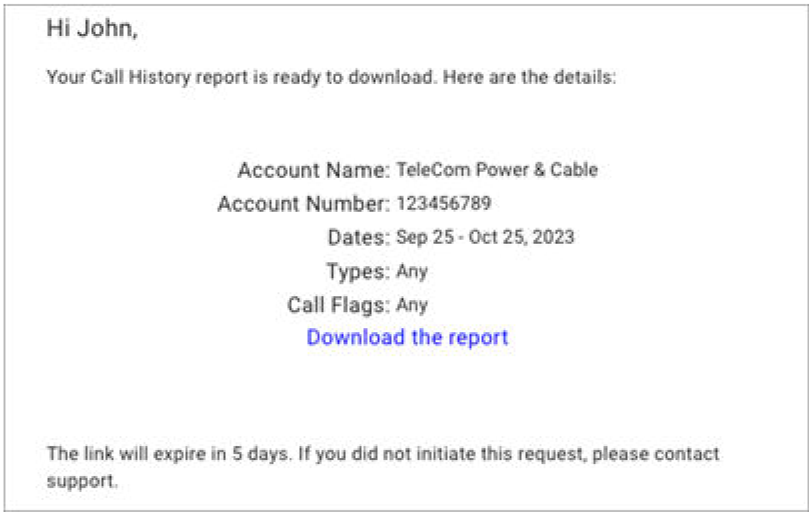
Field	Description
Time and Length	When the call started, connected, and ended, what duration of the call was billed, and the actual length of the call.
Cost	The cost of the call, if the call was within the calling plan, and the rate per minute from the plan.
Origination	The number and location of the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

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:



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 Admin 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.
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.

#	Field Name	Type	Description
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.
S	Cost	double	Rated value of call, according to calling plan product.
T	DialedNumber	string	The digits originally dialed to start the call.

#	Field Name	Type	Description
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."
AH	LegType	string	Indication of direction of call: • Origination • Termination • Forward

#	Field Name	Type	Description
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.

#	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.

#	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