

VoiceConnect

Admin Guide

15 November 2019



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About PSTN SIP trunks

Integration with a PSTN SIP trunk, known as VoiceConnect, provides inbound and outbound calling to and from the global telephone network for your organization. This gives your organization:

- direct-dial numbers for inbound and outbound calling for StarLeaf room systems
- direct-dial numbers for individuals for inbound and outbound calling
- direct-dial numbers for hunt groups in your organization
- telephone numbers that can be set as the caller ID for outbound calls (and these can be set as a site default and/or organization default or per user or room system)
- the ability to choose a specific SIP trunk for outbound PSTN calls on a per site basis

You will use a SIP trunk provider to connect your StarLeaf organization to the PSTN. StarLeaf Cloud supports SIP trunk providers across the globe. The StarLeaf Portal provides pre-configured templates for each supported provider making it simple to configure.

You will need to choose a SIP trunk provider and set up an account with that provider before configuring your StarLeaf organization with the details of that provider and your account.

Note: Direct-dial numbers are also known as DID (Direct Inward Dialing) in the US and DDI (Direct Dial-In) in Europe and elsewhere.

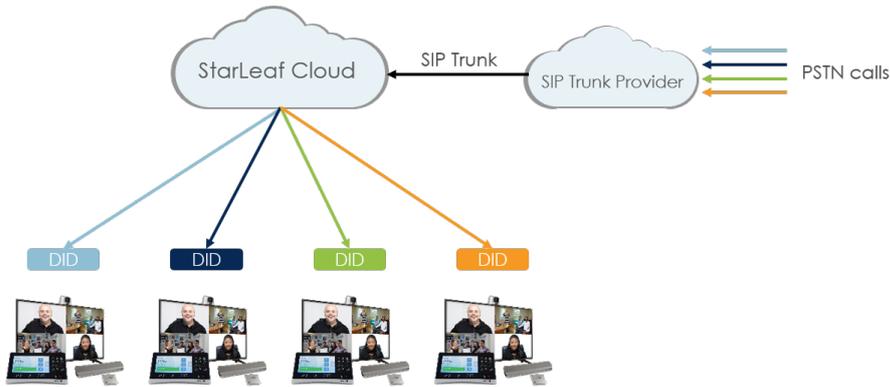
Note: When you configure a PSTN SIP trunk through the Portal, all users in your organization will be able to make PSTN calls from their StarLeaf app. If you have more than one PSTN trunk configured, you can control how a user's calls are routed by the configuration of your sites and trunks.

Integrating PSTN SIP trunks with your StarLeaf organization

It is worth considering how you will integrate PSTN connectivity with your StarLeaf organization before you configure any sites or PSTN SIP trunks. For example, is there more than one geographical location in your organization with different requirements; do you have or need hunt groups; do you have homeworkers calling to the PSTN; do all, or some, users need a direct-dial number? Or do you want to enable StarLeaf room systems for telephone calling?

Example one

This deployment example is an organization that needs to provide PSTN connectivity to StarLeaf meeting rooms.



Every meeting room system is allocated with a direct-dial number. The above diagram shows a small organization with only one site and only one PSTN SIP trunk. However, this is a scalable solution which can be deployed across meeting rooms in multiple sites.

Portal configuration for example one

There is one site configured, the default site:

Sites

Label	Country (Area)	Default	
Default site	United Kingdom	✓	

There is one trunk configured, the Xampleco trunk:

Name	Provider	Site	Status	
Xampleco trunk	Gradwell (UK)	Default site	✓ Online	

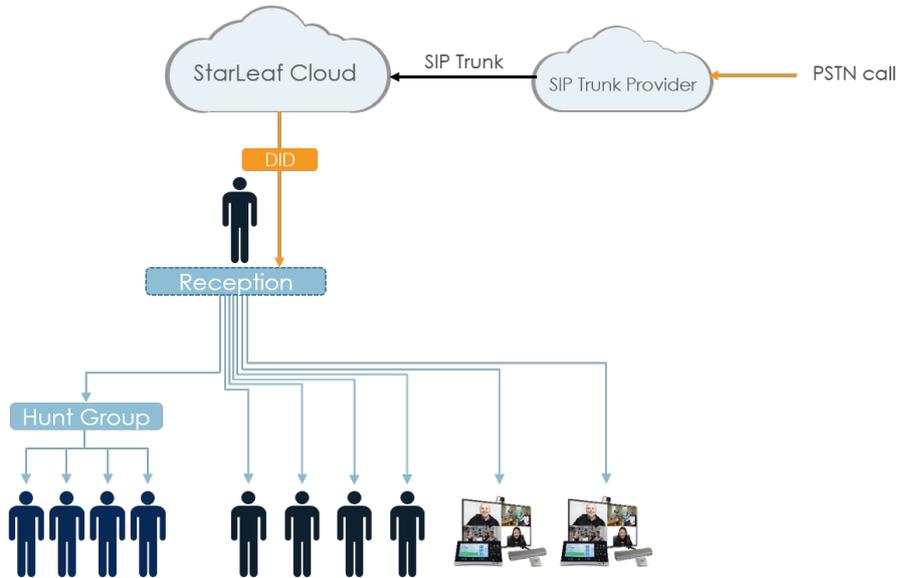
Xampleco have purchased five telephone numbers and assigned one to each of their five room systems:

PSTN SIP numbers

Site	Trunk name	PSTN number	Target	Site default	Organization default	Status
Default site	Xampleco trunk	+44 (0) 12	Amsterdam	<input type="radio"/>	<input checked="" type="radio"/>	✗ Offline
Default site	Xampleco trunk	+44 (0) 123	Utrecht	<input type="radio"/>	<input type="radio"/>	✗ Offline
Default site	Xampleco trunk	+44 (0) 1234	Rotterdam	<input type="radio"/>	<input type="radio"/>	✗ Offline
Default site	Xampleco trunk	+44 (0) 123456	The Hague	<input type="radio"/>	<input type="radio"/>	✗ Offline
Default site	Xampleco trunk	+44 (0) 1234567	Delft	<input type="radio"/>	<input type="radio"/>	✗ Offline

Example two

Example two is a small organization with only one site and only one PSTN SIP trunk. There is a receptionist who has a direct-dial number. The receptionist receives all incoming calls from the PSTN to that direct-dial number and where necessary transfers calls to the relevant person, room system, or hunt group. All outgoing calls to the PSTN from all users will use the same trunk and will use the direct-dial number as the caller ID.



Portal configuration for example two

There is one site configured, the default site:

Sites

Label	Country (Area)	Default
Default site	United Kingdom	✓

There is one trunk configured, the Xampleco trunk:

Name	Provider	Site	Status
Xampleco trunk	Gradwell (UK)	Default site	✓ Online

Xampleco have purchased one telephone number and this is assigned to Richard Cory (the receptionist):

Edit PSTN SIP trunk

Trunk ✓ Online

Provider: Gradwell (UK) Description: One trunk to cover all dialing requirements of my org

Name: Xampleco trunk

> Advanced

PSTN numbers

Site: Default site

PSTN number	Registration name	Extension authentication	Extension password	Assigned
+44 (0) 123456	123	123	123	✓ Richard Cory

The telephone number is also the **Organization default** which means that all outgoing calls (from every user and room system) will use this trunk and have this telephone number as the caller ID.

PSTN SIP numbers

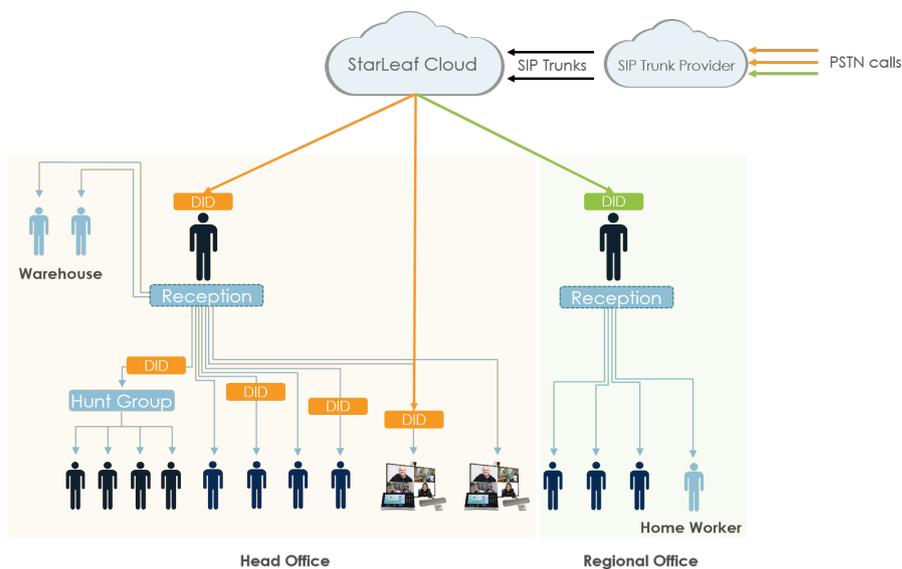
Site	Trunk name	PSTN number	Target	Site default	Organization default	Status
Default site	Xampleco trunk	+44 (0) 123456	Richard Cory	<input type="radio"/>	<input checked="" type="radio"/>	✓ Online

Example three

Example three is a medium-sized organization with a head office, a warehouse, a regional office, and some remote homeworkers.

Direct-dial numbers are allocated to the following:

- The receptionist (Richard Cory) at the head office
- The receptionist at the regional office
- The hunt group
- Two executives
- One room system



Portal configuration for example three

There are two sites configured, the default site (UK) and Scarsdale (US):

Sites

Label	Country (Area)	Default	
Default site	United Kingdom	✓	
Scarsdale	United States (New York)	x	

There are two trunks configured (one for the UK and one for the US):

Name	Provider	Site	Status	
US trunk	Geils (US / UK)	Scarsdale	✓ Online	
Xampleco trunk	MIXvoip (LU)	Default site	✓ Online	

Xampleco have purchased five telephone numbers on the UK trunk and one for the US trunk.

The telephone number is also the **Organization default** which means that all outgoing calls (from every user and room system) will use this trunk and have this telephone number as the caller ID.

Outgoing calls from homeworkers will be routed to the trunk that is configured to be the **Site default** for the US site (the US PSTN SIP trunk in the example), with the caller ID for outgoing calls to be the receptionist's number.

Outgoing calls from the warehouse will be routed to the trunk that is configured to be the **Organization default** (the UK trunk in the example), with the caller ID for outgoing calls to be the receptionist's number.

Note that where a user or room system is not allocated with a direct dial number, any outgoing calls will be routed via the **Site default** if there is one, or the **Organization default**.

Site	Trunk name	PSTN number	Target	Site default	Organization default	Status
Default site	Xampleco trunk	+44 (0) 12	Richard Cory	<input type="radio"/>	<input checked="" type="radio"/>	✓ Online
Default site	Xampleco trunk	+44 (0) 123	Sales team	<input type="radio"/>	<input type="radio"/>	✓ Online
Default site	Xampleco trunk	+44 (0) 1234	Alexis Reena	<input type="radio"/>	<input type="radio"/>	✓ Online
Default site	Xampleco trunk	+44 (0) 123456	Nicodemo Albertus	<input type="radio"/>	<input type="radio"/>	✓ Online
Default site	Xampleco trunk	+44 (0) 1234567	Amsterdam	<input type="radio"/>	<input type="radio"/>	✓ Online
Scarsdale	US trunk	+1 (1) 555	Trudie Doriane	<input checked="" type="radio"/>	<input type="radio"/>	✓ Online

About sites

Configure sites on the Portal for the purposes of location-specific settings such as how StarLeaf will:

- specify the default timezone
- specify the default language to be used on the touchscreen controllers of room systems, StarLeaf app, and other StarLeaf endpoints
- apply location-specific QoS settings in a multi-site deployment
- apply location-specific power-line frequency to eliminate flicker on room systems caused by florescent lights

When used in combination with PSTN SIP trunks, sites also allow StarLeaf to:

- route calls originating from a particular site to a particular PSTN SIP trunk. A site usually maps onto a geographical location, but where necessary you can configure more than one site per geographical location (for example, to allow calls from one geographical location to be routed to more than one PSTN SIP trunk) or you might find that you can group multiple geographical locations into one site if they have the same requirements
- interpret and re-format locally-dialed numbers for the trunk to which they are destined. For example, certain providers expect telephone numbers to be formatted as national numbers whereas other providers expect numbers to be formatted for international dialing
- present a caller ID to the person receiving a call from a PSTN SIP trunk that will make sense to that person within the context of their own location. For example, an incoming call to an endpoint in a US site from a French PSTN SIP trunk will need to look like a French number rather than a locally dialed number

Integration with PSTN SIP trunks (VoiceConnect) is an optional addition to StarLeaf accounts. For more information about PSTN SIP trunks, refer to [About PSTN SIP trunks](#).

How to create a site

Note: When you create an organization, the country you enter will determine your datacenter location. This will also become your default site. You can change the settings of your default site including country, but your organization will not be moved to a different datacenter

Before you create multiple sites for your organization, consider what you are trying to achieve in so doing. Remember that individual users and room systems can be configured to use a different time zone and language to the default for their site or organization.

Broadly speaking, the sites you configure will map onto the geographical locations in your organization. If you are creating sites for the purposes of deploying PSTN connectivity, remember that a site determines the location of any telephone numbers on the trunk. That is, you cannot change the country or area code for a telephone number on a trunk.

For the purposes of configuring PSTN SIP trunks, there is a many-to-one relationship between trunks to sites. You can have many trunks to a site, but a trunk can only exist in one site. It might be practicable to group several of your geographical locations together into one 'site' based on their shared requirements. For example, in some deployments, home workers could be placed into their own site to provide them with the necessary

settings and a default site trunk to which their PSTN calls are routed. However in other deployments, it might be more appropriate to place them into a site along with other physical location where users have the same dialing requirements.

If you have sites enabled on your StarLeaf organization, you will by default have one site pre-configured. This is the default site for your organization. To add additional sites:

1. In the Portal, go to **Sites > Add site**:
2. Complete the fields:
 - **Label**: A name to help you identify this site
 - **Country**: Geographical location of this site
 - **Area**: Some countries are divided into areas for the purpose of PSTN dialing
 - **Default time zone**: The time zone that the site is in. This is for the purpose of displaying the correct time on endpoints and also to schedule conferences at the correct time when participants are in different time zones. You can override the default time zone on a per-user and per-room system basis
 - **Default language**: The default language for users and room systems for this site. You can override the default language on a per-user and a per-room system basis
 - **Power line frequency**: The power line frequency for this organization. Altering this setting can eliminate camera flicker from fluorescent lights. Do not change this setting unless you are experiencing flicker
 - **DSCP value**: Set the DSCP value to be used by StarLeaf room systems, desktop systems, and Phone 2105 in this site. This setting does not affect the StarLeaf app or H.323 systems. Appropriately configured routers can use this DSCP value to prioritize network traffic. This field can be empty if your network does not use QoS. For more information, refer to [Using QoS with the StarLeaf Cloud \(p17\)](#)
3. Click **Apply**.

Add site

Label:
Houston Office

Country:
United States

Area:
Michigan

Default time zone:
[UTC-05] America/Detroit (EST)

Default language:
English (portal and endpoints)

Power line frequency (Hz):
50

DSCP value:

Portal configuration for PSTN SIP trunks

Integration with PSTN SIP trunks, known as VoiceConnect, provides inbound and outbound calling to and from the global telephone network for your organization.

Note: VoiceConnect is an optional extra on your StarLeaf account.

How to configure a PSTN SIP trunk

Step one: Set up an account with a SIP provider

Set up an account with a SIP provider. We support the following providers:

<ul style="list-style-type: none"> ■ Atom IP (UK) ■ Backstage (NL) ■ blueface (IE) ■ CoreDial/Nutopia (US) ■ Deutsche Telefon (DE) ■ DIDLogic (worldwide) ■ Geils (US, UK) ■ Go Trunk (America North) ■ Go Trunk (Europe) ■ Gradwell (UK) ■ Hexanet (FR) ■ Itec (ZA) ■ messagenet.it (IT) ■ MIXvoip (LU) ■ Motto (NL) ■ My Republic (SG) ■ OpenIP (FR) ■ Over the Wire (AU, NZ) ■ OVH (FR) 	<ul style="list-style-type: none"> ■ Paratus (NA) ■ Pure IP (worldwide) ■ sip.us (US) ■ SIPcity (AU) ■ sipgate basic (DE) ■ sipgate basic (UK) ■ sipgate team (DE) ■ sipgate team (UK) ■ siptrunk.com (US) ■ Switch Telecom (ZA) ■ Telavox (SE) ■ Telitcloud (LU) ■ Telnyx (US, CA) ■ TetraVX (US) ■ Twilio (worldwide) ■ Vodafone Iceland ■ Voice flex (UK) ■ Voicehost (UK) ■ Weblink (SE)
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Step two: Configure sites and hunt groups

Consider your requirements and what you are trying to achieve. You might find it useful to refer to [About PSTN SIP trunks](#), [About sites](#), and [About hunt groups](#).

A site determines the location of any telephone numbers on the trunk. That is, you cannot change the country or area code for a telephone number on a trunk.

Step three: Configure the trunk in the StarLeaf Portal

Note: Consider how you want to deploy PSTN if you have multiple sites. Refer to [About PSTN SIP trunks](#); you might want to configure multiple sites in the Portal before you proceed.

1. In the StarLeaf Portal, go to **PSTN SIP trunks** and choose **Add PSTN SIP trunk**:

[PSTN SIP trunks](#)



2. From the Provider drop-down menu, select your SIP trunk provider and complete the following:
 - **Name:** Give the trunk a name (so you can identify this trunk)
 - **Description:** Add a description (optional)
 - **Gateway:** For some providers, the Portal asks for a gateway. This might be a specific domain allocated on a per-customer basis or it will be a region specific gateway and in this case, enter a domain or an IP address
 - **Site:** Choose the site for this SIP trunk. Note that the country (and area, if applicable) of the site will dictate the dialing code for the telephone numbers associated with this trunk
3. Add the telephone numbers that you have bought from the provider. For each number that you have purchased, enter authentication details from your SIP trunk provider. What you enter here varies from provider to provider (and some provider configurations are documented in [PSTN SIP authentication](#)). You might need to contact StarLeaf Support for help.
 - **Registration name:** Sometimes this is the telephone number, a username, or an account number and sometimes this is not required
 - **Extension authentication:** You must enter a value here. It will be the same as the **Registration name** (unless that is empty). Often it is the username or the telephone number, but sometimes it is an account number
 - **Extension password:** The password for the telephone number

	PSTN number	Registration name	Extension authentication	Extension password	Assigned
	+44 (0) 123	xyzxyz	xyzxyz	wxyzwxyz	None

4. Notice that the numbers are currently not **Assigned**. Complete the next step.

Step four: Assign the PSTN numbers

1. In the StarLeaf Portal, go to **PSTN SIP numbers**. You will see all the numbers that you have purchased and added to your configured trunks. For each PSTN number you can configure these settings:
 - **Target:** Choose a user, room system or hunt group to be the target for a PSTN number. This means that any incoming calls to this number will be routed to the chosen user, room system or hunt group. It also means that outgoing calls from the target, where that is a user or a room, will use the PSTN number as the caller ID. Note that members of hunt groups will not use the target as their outgoing caller ID and will instead use either their own target number if they have one, or the site default if there is one, or the organization default
 - **Site default:** You can choose one PSTN number per site to be the site default. This means that all outgoing calls from the site will use this number as their caller ID (unless the room or user making the call is a target for another PSTN number)
 - **Organization default:** You can choose one PSTN number to be the organization default. This will be used as the caller ID where the user or room both does not have

a site default *and* is not a target for another PSTN number

PSTN SIP numbers

Site	Trunk name	PSTN number	Target	Site default	Organization default	Status
Default location	Head Office DIDs	+44 (0) 12		<input type="radio"/>	<input type="radio"/>	✓ Online
Default location	Head Office DIDs	+44 (0) 123		<input type="radio"/>	<input type="radio"/>	✓ Online
Default location	Head Office DIDs	+44 (0) 1234		<input type="radio"/>	<input type="radio"/>	✓ Online
Default location	Head Office DIDs	+44 (0) 123456	Richard Cory	<input type="radio"/>	<input checked="" type="radio"/>	✓ Online
Default location	Head Office DIDs	+44 (0) 1234567		<input type="radio"/>	<input type="radio"/>	✓ Online
Scarsdale	Scarsdale US office and homeworkers	+1 (1) 98988888	Alexis Reena	<input checked="" type="radio"/>	<input type="radio"/>	✓ Online

[Apply](#)

2. Click **Apply**.

Note: You can also assign PSTN numbers through the [Users > Edit](#), the [Room systems > Edit](#), and the [Hunt groups > Edit](#) pages.

Note: Assigned PSTN numbers do not automatically appear in the directory, however if necessary, you can add these by editing the details for a room, user, or hunt group. For example, you can add the PSTN number as a user's **Work** number.

SIP trunk integration guides

For VoiceConnect guides that are specific to certain SIP providers, select one of the following:

- [sip.us](#)
- [Telnyx](#)
- [Twilio](#)

About hunt groups

Note: Hunt groups can be configured through the Portal. This is an additional option. Contact StarLeaf technical support or your reseller if your organization requires hunt groups.

Hunt groups allow incoming calls to a specific number to be directed to a defined group of extension numbers (belonging to the hunt group members). When a call comes into the hunt group, the apps (or endpoints) of the members will either ring simultaneously or in a sequence depending on the configuration of the hunt group.

To view, create, and edit hunt groups, in the Portal go to **Hunt groups**.

For every hunt group, specify the type of hunt group, the ringing duration, the fallback type, and whether or not the hunt group will cause StarLeaf on mobile devices to ring.

The Cloud provides five types of hunt group:

- **Linear:** An incoming call to the hunt group will ring the first member of the hunt group and if there is no answer within the **Ringing duration** time (a configuration option), the call will be routed to the second member of the hunt group and so on until the call is answered. If nobody in the hunt group answers the call, the configured **Fallback** will be applied
- **Group:** An incoming call to the hunt group will ring every member of the group simultaneously. If nobody in the hunt group answers the call, the configured **Fallback** will be applied
- **Whole organization:** An incoming call to the hunt group will every user and room system in the organization simultaneously. If nobody answers the call, the configured **Fallback** will be applied
- **All users:** An incoming call to the hunt group will ring every user in the organization simultaneously. If nobody answers the call, the configured **Fallback** will be applied
- **All room systems:** An incoming call to the hunt group will ring every room system in the organization simultaneously. If nobody answers the call, the configured **Fallback** will be applied

If nobody answers a call to a hunt group, you can configure a fallback which is one of:

- **None:** the call will hang up if nobody answers it
- **User/room system/hunt group:** the call will be diverted to a particular user, room system, or hunt group
- **Voicemail:** the call will be diverted to the voicemail system of a particular user

Create a hunt group

1. In the Portal go to **Hunt groups > Add hunt group**.
2. Type a **Name** for the hunt group.
3. Choose whether or not the hunt group will cause StarLeaf on mobile devices to ring. For example, for some hunt groups it will only be appropriate for members to answer if they are at their desk on their desktop device.
4. Choose the **Hunt type** (see above for more details of hunt types).
5. Type the **Ringing duration**. This is the amount of time for which an app or endpoint will ring before the call is routed to the next recipient (if this is a linear group) or to the configured fallback.

6. If this is a **Linear** or **Group** hunt group, add the **Members**. These can be users or room systems.
7. Choose a fallback type:
 - **None**: the call will hang up if nobody answers it
 - **User/room system/hunt group**: the call will be diverted to a particular user, room system, or hunt group that you enter
 - **Voicemail**: the call will be diverted to the voicemail system of a particular user that you enter
8. Choose **Apply** and you will see that the **Dialing information** is automatically completed by the Portal. For more information about video addresses, refer to [About StarLeaf video addresses](#).

Add hunt group

▼ Hunt group settings

Name

Sales team



Ring StarLeaf app on mobile devices

▼ Members

Hunt type

Group



Ringing duration (seconds)

20

Members

Richard Cory



Nicodemo Albertus



Alexis Reena



Fallback type

None



> Dialing information

salesteam@xampleco.call.sl, +44 (0) 3300571888 (Xampleco SIP Trunk)

PSTN SIP authentication

For a complete list of supported SIP trunk providers, refer to [Portal configuration for PSTN SIP trunks](#).

The information below provides details about which information to use for authentication of telephone numbers:

Atom IP (UK)

StarLeaf Portal	Service provider
Registration name	Username
Extension authentication	Username
Extension password	Password for the telephone number
Gateway	Not required by StarLeaf Portal

DIDLogic (US)

StarLeaf Portal	Service provider
Registration name	SIP user name
Extension authentication	SIP user name
Extension password	Password for the SIP account
Gateway	Not required by StarLeaf Portal

Geils (US/UK)

StarLeaf Portal	Service provider
Registration name	Leave blank
Extension authentication	Telephone number
Extension password	Password for the telephone number
Gateway	Not required by StarLeaf Portal

GoTrunk (Europe)

StarLeaf Portal	Service provider
Registration name	SIP endpoint > SIP username
Extension authentication	SIP endpoint > SIP username
Extension password	SIP endpoint > SIP password
Gateway	Not required by StarLeaf Portal

Gradwell (UK)

StarLeaf Portal	Service provider
Registration name	Account number
Extension authentication	Account number
Extension password	Password for the telephone number
Gateway	Not required by StarLeaf Portal

messagenet.it (IT)

StarLeaf Portal	Service provider
Registration name	Username
Extension authentication	Username
Extension password	Password
Gateway	Not required by StarLeaf Portal

Using QoS with the StarLeaf Cloud

Quality of Service support is an optional extra that can be included as part of your organization's endpoint subscriptions. When enabled for an organization, you can set a DSCP value for that organization. This will set the DSCP bits in the signaling and media packets from StarLeaf room systems, desktop systems and the Phone 2105. It will not affect traffic from StarLeaf apps or H.323 endpoints.

For organizations with Quality of Service enabled, you can find the setting in the Portal. From the organization page in the Portal go to **Sites > Edit site > DSCP value**.

Edit site

Label:
Default site

Country:
United Kingdom

Default time zone:
[UTC] Europe/London (GMT)

Default language:
English (portal and endpoints)

Default max send rate (kbit/s):
Automatic

Default max receive rate (kbit/s):
Automatic

Power line frequency (Hz):
50

DSCP value:
46

OK Apply

About StarLeaf video addresses

How StarLeaf assigns video addresses

Within an organization, every user and room system has a directory number and is also listed in the directory.

StarLeaf users and StarLeaf guests can call each other by email address and this works between organizations.

Every user that is provisioned is also automatically assigned a unique video address such that they are reachable from any standards-based video endpoint that is routable on the Internet.

To view a user's video addresses, go to the [Edit user](#) page for that user and view the [Dialing information](#). An example of what this might look like:

Dialing information	
StarLeaf video address	alexis.reena@exampleco.com
External video address	<input type="text" value="alexis.reena"/> @exampleco.call.sl <input checked="" type="checkbox"/> autocomplete
Extension	2101
Legacy video address	830652101@exampleco.call.sl

Address types:

- StarLeaf video address:** the user's email address. StarLeaf users and StarLeaf guests can call each other by email address
- External video address:** StarLeaf provides external addresses for each user, room system, and scheduled meeting. An external video address is to be used by people outside the organization who are using H.323 devices or SIP devices or Skype for Business clients that are routable on the Internet. By default, StarLeaf autocompletes this address. However, if you require, you can edit the first part of the address. When a user sends a guest invite, all dialing information for this user is also provided for the guest. This means that a guest can dial from existing video conferencing equipment if he prefers. A user's external video address is displayed on the [settings > about](#) screen of the StarLeaf touchscreen controller and Breeze
- Extension:** StarLeaf assigns each user an extension number. Other StarLeaf users inside your organization can either choose to call someone from the directory (contacts), or they can dial the extension of the person they want to talk to. In the above example, the extension is 1100
- Legacy video address:** this address is only necessary where someone is attempting to dial this endpoint from an older H.323 device where it is not possible to dial certain characters. StarLeaf creates this address from the directory number and a unique organization prefix that is assigned when a StarLeaf account is created in combination with the account **Subdomain**

Examples

An example user is configured with these settings:

Organization name: Xampleco

Subdomain: xampleco

Organization prefix: 83065

User first name: Alexis

User last name: Reena

User email address: alexis.reena@xampleco.com

User extension: 2101

StarLeaf assigns these addresses for Alexis Reena:

StarLeaf video address: alexis.reena@xampleco.com

Legacy video address: 830652101@xampleco.call.sl

Who can you call and who can call you?

Your StarLeaf endpoints are connected to the StarLeaf platform. This means that from a StarLeaf endpoint, you can call anyone who has a standards-based video device connected to the public Internet and they can call you. If you want to video-call someone who does not have a StarLeaf device, send that person a guest invite from the StarLeaf app or a StarLeaf room system.

Depending on your account settings, you might also be able to call out to the public telephone network.

The features and functionality of StarLeaf endpoints are fully explained in the documentation available in the [Knowledge Center](#).

Legal information

Third party software acknowledgments

Acknowledgments of third-party software are available at:

www.starleaf.com/support/legal

Disclaimers and notices

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