Call2Teams®

Getting your PBX ready for Call2Teams

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1 Introduction

Call2Teams is a service that sits between your PBX service and Microsoft Teams on Office 365. The service is designed to connect to your PBX using the standard 'SIP Registration' method, in the same way as with a standard VOIP desk phone e.g. Polycom or Yealink.

This standard approach means that virtually all PBX systems can work with Call2Teams. In practice, there may be changes required to your PBX or network to create a successful connection. This document sets out the process you can run through to configure and validate your PBX connection so that Call2Teams can connect successfully.

1.1 About your PBX

Your phone system is likely to either be

- a hosted service, operating in the datacenters of your provider and already accessible from everywhere, or
- an on-premise system, probably physically located in your offices and accessible from the users in your building.

Hosted PBX services are normally ready-to-go and require little intervention; on-premise systems typically need a little more work.

You should consult your service provider or PBX maintenance company for guidance to create and manage the SIP accounts needed for Call2Teams to connect. This document may also be useful to them.

1.2 The audience for this document

This document is provided for customers to understand and configure their PBX for access by the Call2Teams service.

It is expected the customer will be able to call upon the following resources and experience:

- Networking and firewall configuration
- PBX configuration and management
- SIP device configuration (phone/softphone)
- Microsoft Teams basic configuration (PowerShell skills are not required)

Customers often need to involve the organization or person that maintains and configures their phone system.

1.3 The configuration process

The process of configuring your PBX for use with Call2Teams is split into three discrete elements:

1. Enabling remote access (without VPN) to your PBX
2. Test the remote access of the PBX
3. Call2Teams interop with your PBX & testing

2 Enabling remote access to your PBX

Call2Teams connects to your phone system from the Internet. You should configure access to your phone system in the same way as you would for mobile or home workers using a standard SIP softphone (e.g. MicroSIP, CounterPath, Zoiper) to connect to the PBX. If you are using a cloud PBX service then there is probably nothing for you to do here, but if you have an on-premise PBX then there may be a configuration change to your router or firewall to allow the access.

It is outside the scope of this document to go into the configuration of various firewalls and routers but here are a few guidelines for your IT admin to configure the firewall.

Where it is already understood that the PBX is readily accessible from the internet this section can be skipped, go to section 4

2.1 Important notes to Network Administrators:

- If your firewall supports SIP ALG, we strongly recommend disabling this.
- Certain network routers and switches may have a problem handling fragmented UDP packets. Part of this readiness process will test SIP UDP fragmentation and issues may need to be addressed or the TCP protocol used.
  If the PBX is hosted in Azure or Azure Stack then it will not cope with fragmented UDP so skip straight to TCP or TLS.
- Although Call2Teams uses a relatively static range of IP addresses, all registration and SIP for all users will emanate from one or two IP addresses, available in the portal. Media will come from a wider range of addresses
- VPN access to your network is not available for the service.

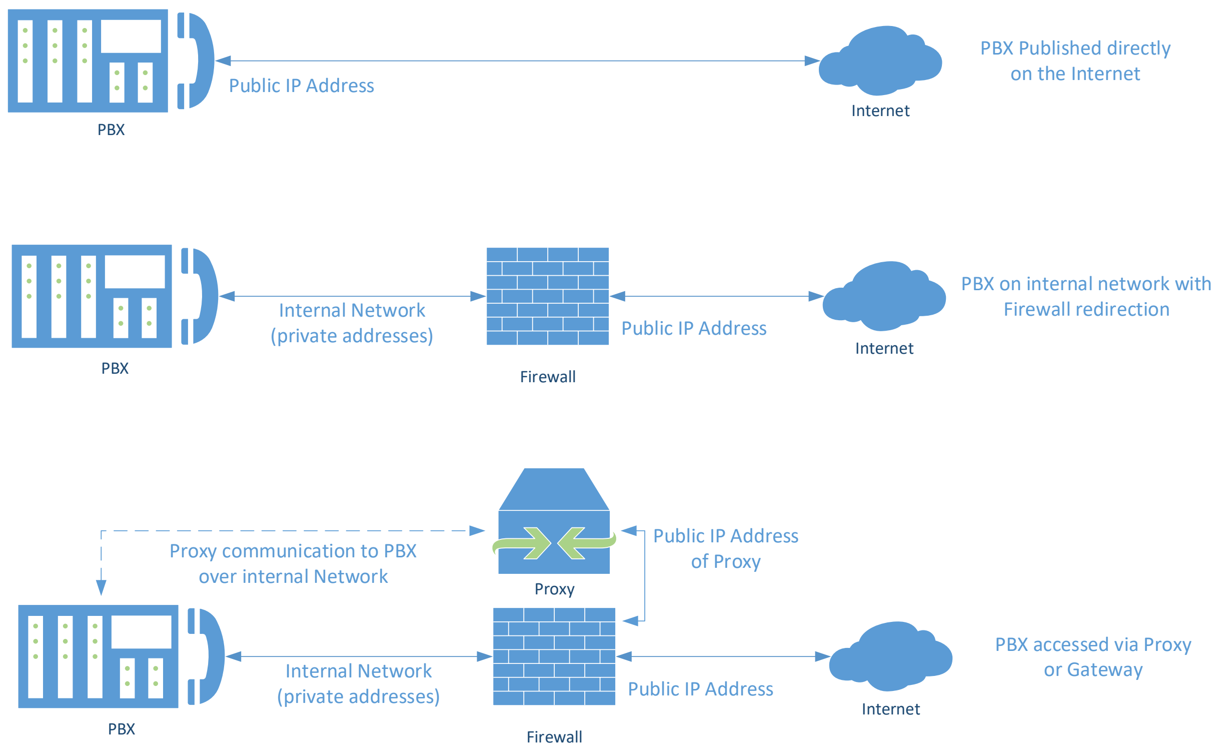
2.2 Do you have an SBC or SIP proxy/gateway?

Some on-premise PBXs are feature limited to restrict external connection from the internet. The vendors of these platforms will mandate the use of Session Border Controller (SBC) or Border Gateway. For this document, we will refer to this family of appliances or services as 'proxy'

Proxies sit at the edge of a customer network and relay voice traffic between the public internet and the PBX. They provide additional security and management for external connections, acting as a 'voice firewall' and can translate between different variants of the SIP protocol.

If you have a proxy on your network, then this will need configuring to allow external SIP connection from our service in the same way as a firewall.

2.3 Connection topology


PBX
Public IP Address
Internet
PBX Published directly on the Internet
PBX
Internal Network (private addresses)
Firewall
Public IP Address
Internet
PBX on internal network with Firewall redirection
Proxy communication to PBX over internal Network
Proxy
Public IP Address of Proxy
PBX
Internal Network (private addresses)
Firewall
Public IP Address
Internet
PBX accessed via Proxy or Gateway

There are three main methods of accessing a PBX from the internet:

1. Direct publishing: The PBX has a public IP address on its interface and devices can directly access the PBX by IP address or hostname.

2. Accessed via firewall redirection: The PBX is on an internal network and a firewall or router is performing Network Address Translation (NAT) between the public internet address and the internal network addresses. In this case, certain traffic types from the internet will be allowed through and redirected to the PBX to allow internet devices to talk to the PBX through the firewall, in the same way as a directly published PBX.

3. Accessed via SBC, proxy, or gateway: The Proxy is directly accessible from the internet (normally managed by the firewall). The proxy handles the direct conversation with the connected devices and has a separate communication link with the internal PBX, often using proprietary protocols.

2.3.1 Direct published PBX

For PBXs that are directly published, this is seldom a problem as they are by default accessible from internet-based users.

2.3.2 PBX behind firewall

PBXs behind a firewall will require specific redirections to be set-up to allow connections from internet-based devices to traverse the firewall and be presented to the PBX. Typically, there are two sets of redirections required:

SIP Signaling: The standard for this is Protocol = UDP, Port = 5060, but can vary.

MEDIA: This port range is highly dependent on the PBX configuration, but for example, 3CX PBXs use Protocol = UDP, Port range = 9000-10999.

An informative in-depth article on NAT and firewalls can be found here: https://www.3cx.com/blog/voip-howto/firewall-nat-pat-stun/

2.3.3 Proxy or gateway

The role of the proxy is to publish the PBX services directly to the internet. Documentation will be provided by the vendor of the proxy for the setup to allow internet-based devices to connect.

2.4 Encryption

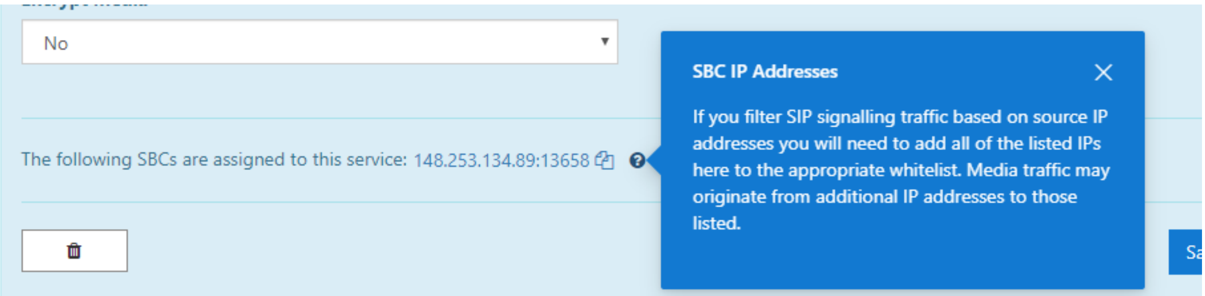
For customers wishing to use encrypted SIP and media, the PBX will require a real SSL certificate (not self-signed) and the renewal of this certificate should be managed to avoid future service disruption.

Be sure to publish the required intermediate certificates as well as the main certificate. You can check the SSL publishing using an online SSL checker site such as sslchecker.com or decoder.link.

2.5 Firewall rules

If you wish to restrict public access to the PBX then there are two components:

For signaling, the IP addresses and port numbers are shown in the Call2Teams portal once the PBX service has been set up. The firewall should allow traffic from any port on those IP addresses using the protocol that corresponds to the chosen transport protocol (udp for "UDP" and tcp for "TCP" or "TLS").



The following SBCs are assigned to this service: 148.253.134.89:13658

SBC IP Addresses

If you filter SIP signalling traffic based on source IP addresses you will need to add all of the listed IPs here to the appropriate whitelist. Media traffic may originate from additional IP addresses to those listed.

For media, this should be open to any IP address and port. Consult the PBX manual to determine which ports it uses for media and allow traffic from any source address.

Note: During the test phase (Section 3) add & allow the public IP of the softphone client you are using.

2.6 PBX specific remote access considerations

Some PBXs, particularly on-premise PBX platforms require additional configuration or licensing to support remote access.

2.6.1 Does your PBX need additional licenses for remote access?

Several PBX manufacturers require an additional license to support a generic 3rd party SIP phone. This is common where the PBX is built to operate with proprietary desk phones and devices. It is important to check and understand this requirement and procure the needed licenses for remote access for each user that will be connected to Microsoft Teams.

2.6.2 Does it need additional licenses for users to have multiple SIP devices (desk phone and Call2Teams)?

Where a user will retain their desk phone and have Teams operating alongside, the user will need two devices attached to their extension. Some PBXs need additional licenses for this arrangement (sometimes called 'shared call appearance') so where a user does need multiple devices, this license requirement should be checked. Note: A user may have several Teams devices connected to the PBX via Call2Teams, this will only use one device connection to the PBX, so just one device should be provisioned and licensed for the Teams user.

2.6.3 Does it use different ports for remote users?

Some PBX manufacturers use a different network port number for remote user connections, so don't assume that the port number your desk phone connects to is the same as the port you need to publish for remote user access.

2.6.4 Does the PBX have ACLs in addition to the firewall?

If the PBX has the built-in capability to restrict the network ranges it will allow connections from (Access Control Lists), then this should be set to allow the external addresses required for the test phase and for Call2Teams to connect.

2.6.5 Public IP presentation

When handling SIP connections to the outside world, it's important that the messages from the PBX refer to the public IP address of the PBX, or the hostname, and not the private LAN address. This way traffic coming back to the PBX will know the destination address of the PBX.

For the UDP or TCP protocols, the public IP address should be used for both SIP and media.

For the TLS protocol, the full external hostname should be used for SIP, which must match the SSL certificate, and the IP address should be used for media.

If the PBX supports remote access, then it may be labeled as 'external address' or similar. If the PBX is using a proxy or SBC, this is likely to be a configuration parameter on that device.

2.6.6 Registration timeout

SIP devices need to re-register to the PBX from time to time to ensure the connection is maintained to receive incoming call notifications. Call2Teams works best with a long timeout and the recommended timeout setting is 3600 seconds (one hour). Registration timeout is normally dictated by the PBX and where this can be set, it should not be lower than 600 seconds (10 minutes) for Call2Teams.

2.6.7 SIP user agent

Some PBXs restrict the devices that can attach by the identity contained within some of the messaging produced by the device. A specific string call 'SIP user agent' commonly identifies the brand/model/revision of the connecting device and a PBX may be configured to only accept connections from certain settings. Ensure 'Call2Teams.com' is in an allowed list (also for testing allow the 'MicroSIP/3.17.7' user agent – version number may be different, see the log file)

3 Test access to your PBX

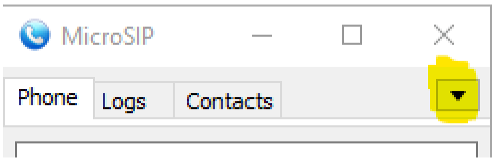
Following the setup as detailed previously, the PBX will be reachable from outside of the organization network, we recommend testing the access to your PBX with a generic tool to verify this.

Here is the process you can use to test your network and PBX for external access and the readiness for Call2Teams

1) Download the *MicroSIP Lite* softphone from www.microsip.org and install it on a PC
2) Connect the PC using MicroSIP on an **external IP address** (i.e. from a PC outside of your network, perhaps by tethering to a mobile hotspot). This is essential to verify the connection can be made through your firewall or Proxy, from the outside to the inside.
3) Configure the MicroSIP account to connect to your PBX with default settings. Use the UDP protocol, unless you have a high-security requirement and have set your PBX up for TLS.

3.1 Configuring MicroSIP

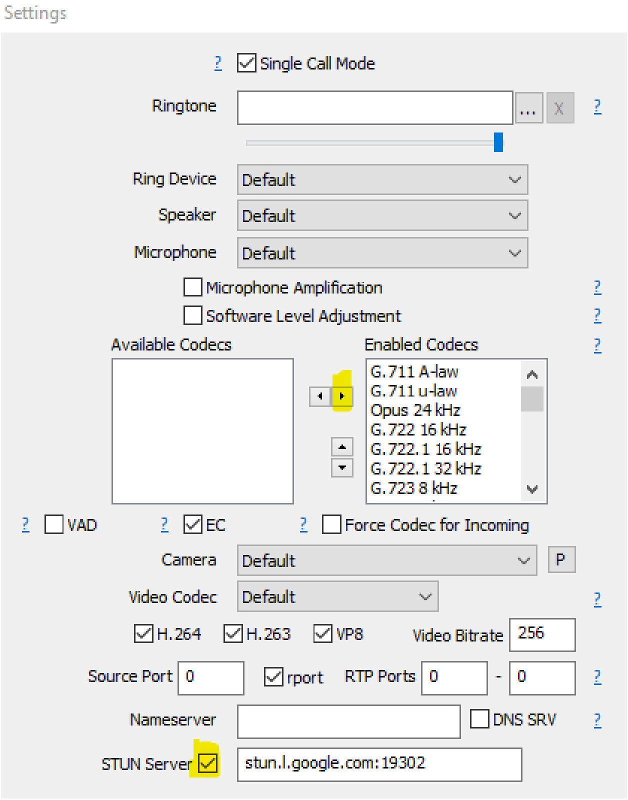
Click on the menu button and select *Settings*.



Move all the codecs from the *Available Codecs* list to the *Enabled Codecs* list and check the *STUN Server* option.

Using all the codecs makes the network packets larger so that UDP fragmentation support can be validated, and using STUN means that MicroSIP will use the public IP address when making calls, which more closely matches how the Call2Teams service works.

Save the settings.



Settings

☑ Single Call Mode

Ringtone

Ring Device: Default

Speaker: Default

Microphone: Default

□ Microphone Amplification

□ Software Level Adjustment

Available Codecs

Enabled Codecs:
G.711 A-law
G.711 u-law
Opus 24 kHz
G.722 16 kHz
G.722.1 16 kHz
G.722.1 32 kHz
G.723 8 kHz

□ VAD ☑ EC □ Force Codec for Incoming

Camera: Default

Video Codec: Default

☑ H.264 ☑ H.263 ☑ VP8 Video Bitrate 256

Source Port 0 ☑ rport RTP Ports 0 - 0

Nameserver □ DNS SRV

STUN Server ☑ stun.l.google.com:19302

Click on the menu button and select *Add Account*.

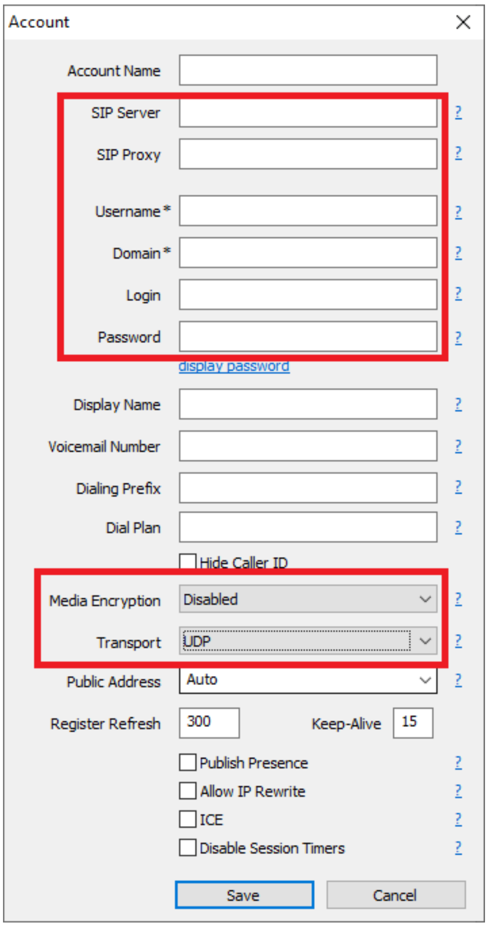
Edit the MicroSIP account settings to create a successful registration of the MicroSIP client. The settings will be required in the fields outlined on the account form above until the online status shows green and online (if you see green with online 'Outgoing', this is not fully set up).

There are many permutations for configuring the SIP client, and this will vary from PBX to PBX. Often, it's helpful to reference guides for setting up other SIP devices for the PBX to understand how the SIP config on the PBX has been designed.

Achieving successful external registration for the PBX is essential for the Call2Teams service to operate.

Once the registration is successful, verify calls can be made and received using the MicroSIP client.

Having a successful MicroSIP account profile will be very helpful when understanding the fields in the Call2Teams portal where various entries are made for your PBX.



Account

Account Name	
SIP Server	
SIP Proxy	
Username*	
Domain*	
Login	
Password	
	display password
Display Name	
Voicemail Number	
Dialing Prefix	
Dial Plan	
	□ Hide Caller ID
Media Encryption	Disabled
Transport	UDP
Public Address	Auto
Register Refresh	300
Keep-Alive	15

□ Publish Presence
□ Allow IP Rewrite
□ ICE
□ Disable Session Timers

4 Call2Teams configuration to work with your PBX

When the PBX is enabled for remote access and it is understood how to create SIP credentials for users, the Call2teams service needs to be configured to connect to the PBX.

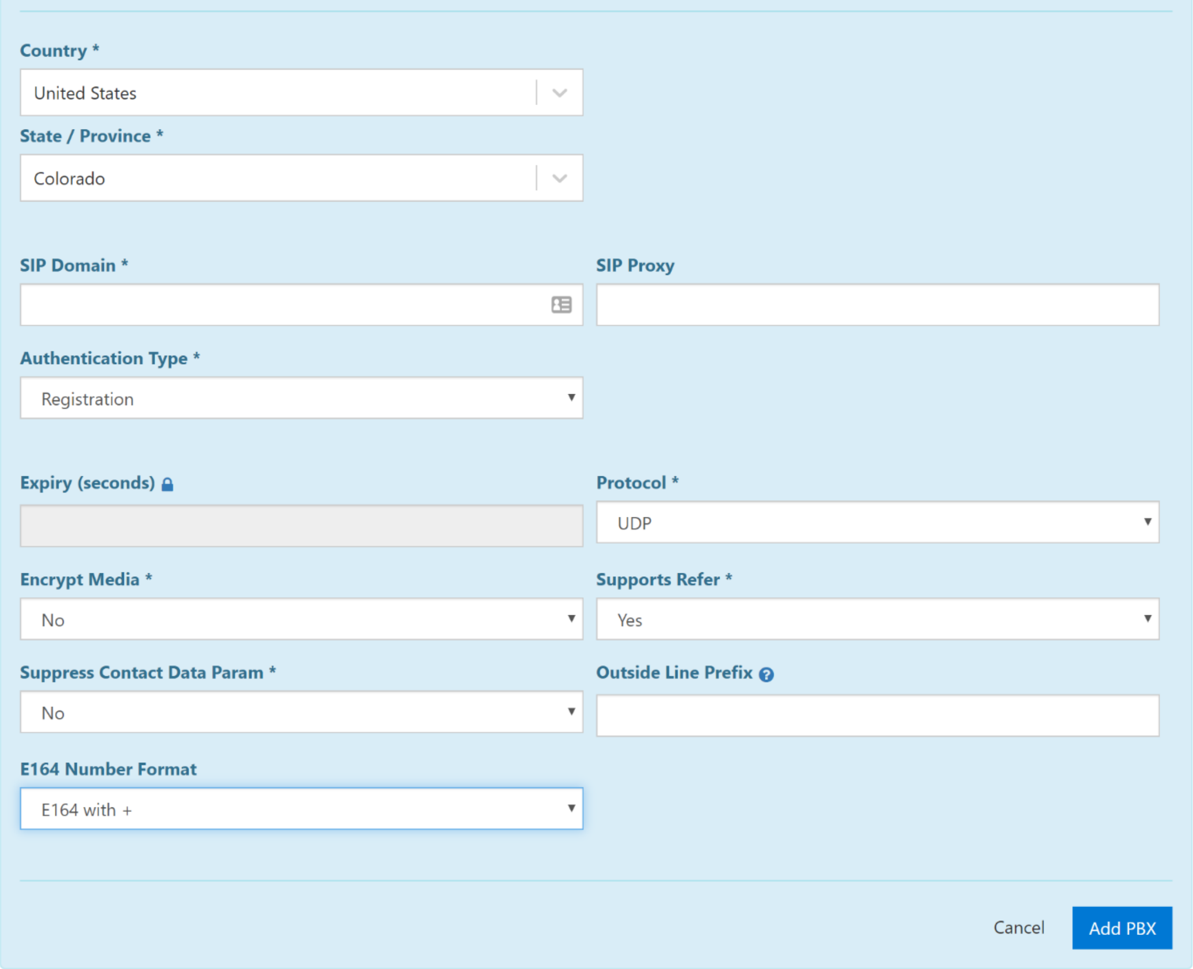
This requires two areas of configuration:

- A service-level configuration to define how every user connects to the PBX
- Per-user configuration to allow each user to provide a device registration to the PBX

4.1 Configure the connection to your PBX

In the Call2Teams portal, the services tab allows for the overall configuration between Teams and the PBX.

When selecting the PBX option, there will be a list of available PBX types or services. Select the most appropriate template for your PBX, if the PBX is not listed, select 'Custom' or 'Not listed'. This will bring up all available options for the PBX connection. If you've selected a defined PBX template, then a subset of these fields will be available in the configuration form.



Country *
United States

State / Province *
Colorado

SIP Domain *

SIP Proxy

Authentication Type *
Registration

Expiry (seconds)

Protocol *
UDP

Encrypt Media *
No

Supports Refer *
Yes

Suppress Contact Data Param *
No

Outside Line Prefix

E164 Number Format
E164 with +

4.1.1 How to complete the PBX configuration form

The following field descriptions will assist you in configuring the PBX settings form

4.1.1.1 Country & State/Province

Call2Teams has servers spread globally across multiple continents. Enter the location of PBX here to ensure the most appropriate service location is configured as the primary note for the PBX to connect to. This ensures the best call quality. The service will configure two nodes for each PBX to ensure high availability.

4.1.1.2 SIP Domain

This is usually just the IP address or public hostname used to reach the PBX. If your PBX requires you to use a specific value here, then do so and specify the IP address or public hostname in the SIP Proxy field. Do not enter a port number on the SIP Domain

4.1.1.3 SIP Proxy

This is usually left blank but can be used to specify the IP address or public hostname to reach the PBX if the PBX requires the SIP domain to be set to a different value.

A port number can be specified by appending a colon and the number to the SIP Proxy field.

4.1.1.4 Authentication Type

This should be set to 'Registration' unless you are deploying an advanced configuration and have been advised otherwise.

4.1.1.5 Protocol

'UDP' is the most common protocol to use, and the one most likely to be supported particularly on older PBXs. However, if the network equipment does not handle fragmented UDP packets then 'TCP' could be used.

If end-to-end encryption is required, then select 'TLS'.

4.1.1.6 Encryption Media

This should be set to 'no' unless encryption is required, and the 'TLS' protocol is configured.

4.1.1.7 Supports Refer

In most cases, this can be set to 'Yes'. Only if the PBX does not have support for transferring calls should this be set to 'No'.

4.1.1.8 Suppress Contact Data Param

If you have a single-organization PBX using a single public IP address, then you should set this to 'No' as the alternative can hinder correct function with some PBXs.

If you have multi-tenant or multi-server PBXs then try setting this to 'Yes' and if it causes an issue, then revert to 'No'.

4.1.1.9 Outside line prefix

If the PBX required a number to be dialed to reach an outside line, enter this here and it will be automatically prepended; this means Teams users do not need to specify this when dialing and can simply dial the destination PSTN number.

4.1.1.10 E164 Number Format

Teams uses E164 numbering, which is where numbers begin with a '+' followed by the country code.

If the PBX expects localized numbers, then you can specify the mapping here.

e.g. if the PBX expects in-country numbers to start with a '0' and international numbers to start with '00' followed by the country code then enter '0' as the national dialing prefix and '00' as the international prefix.

5 Creating SIP accounts on your PBX

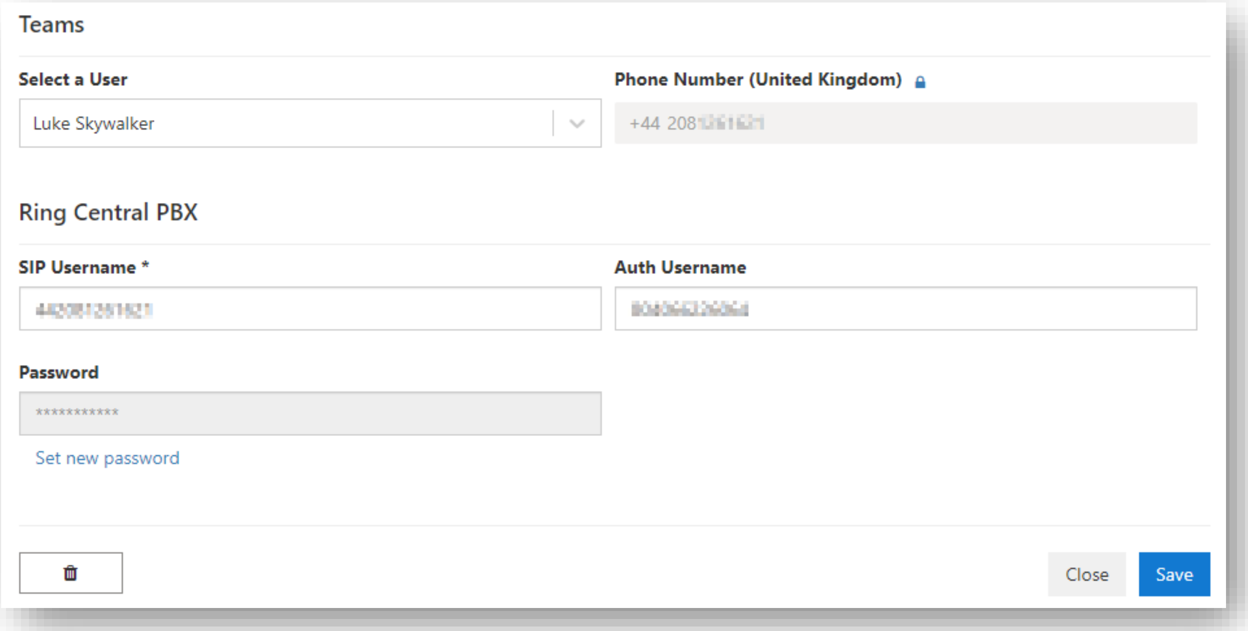
Typically SIP credentials will contain the following information. PBX systems often refer to SIP *devices* rather than *users*. This is because a user on a phone system may have several devices (e.g. desk phone and soft phone) that will be connected to their user extension:

Item	Description
SIP Username	A per-user or per-device identity. Occasionally the same as the extension number.
SIP Auth Username	Not always required; if SIP username is the user identity then SIP Auth Username is the specific device identity, often more complex than SIP username and not memorable. If this field is not required, then it can be left blank
SIP Password	A per-device/user password setting

The per-user settings are entered into the Call2Teams service 'Users' tab to map each Microsoft Teams user to the SIP device credentials created.

A phone number is also required, this may be a full DID number or just an extension. The number must be unique for each user. The country code for the user is driven from the user's country setting in the Office 365 license area – if it is incorrect then alter it in the Office 365 portal and then run another sync into Call2Teams.

SIP Username should not include an @ symbol.



Teams

Select a User: Luke Skywalker

Phone Number (United Kingdom): +44 208[redacted]

Ring Central PBX

SIP Username *: [redacted]

Auth Username: [redacted]

Password: ***********

Note: Remember to click sync again once this user configuration is done, as this will perform the Office 365 configuration for those users.

5.1 Additional configuration topics

5.1.1 Dialing Limitations

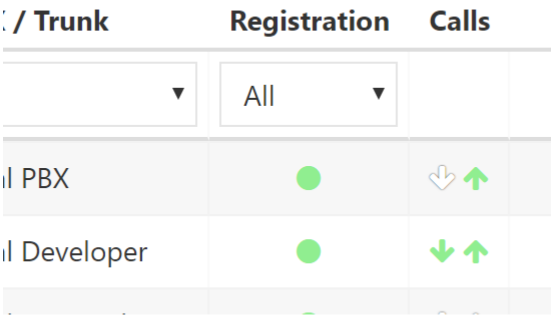
There are some dialing rules built into Teams that cannot be circumvented:

- Numbers beginning with '0' are converted to full E164 numbers. If you have extension numbers beginning with '0' then consider changing them so they don't have this prefix
- Many PBXs have feature codes beginning with '*' that can be dialed. Most will work from Teams, but *11, *12, and *13 do not. Microsoft has indicated that they reserve the right to block some other two-digit star codes as they allocate them for new native Teams features, although at the time of writing there is no sign of this. If you make use of these blocked feature codes then look to see if the PBX can be configured to change the codes to ones that will not be blocked by Teams.

5.2 SIP registration indicators and error messages

SIP registration status is shown against users in the Users tab. This indicator will show green if registration is operating normally.

If the indicator is red, the brief reason will be displayed when you hover the mouse pointer over the indicator. The message shown varies between PBX vendors, and some of the more common ones are shown in the table below:



/ Trunk	Registration	Calls
▾	All ▾	
l PBX	●	↓↑
l Developer	●	↓↑

Message	Advice
Unable to resolve URI	Either the SIP Proxy (if specified) or SIP Domain is incorrect
No response received after xx seconds	The PBX did not respond. Check the firewall and the SIP Proxy/Realm. Specify a port in the SIP Proxy if required.
Authentication Failure	Usually means that the SIP Username, SIP Auth Username (if specified), or SIP Password are incorrect. It could be that the SIP Realm is incorrect if the PBX checks this.
Declined	
Forbidden	
Invalid password	
Not found	
Proxy Authentication Required	
Unauthorized	
User unknown	

6 Initial confidence tests

Perform the following tests to verify compatibility with your PBX for standard confidence levels:

1. Ensure there are at least 2 users registered
2. Test users can call out to the PSTN from Teams for a call duration of at least 1 minute
3. Test users can call the extension number of other PBX users from Teams
4. Make calls (longer than 1 minute) to Teams users to their DDI from the PSTN
5. Make calls into Teams from the PSTN then have a user transfer the call back out to a PSTN number

7 Further testing for full interoperability validation

The following tests provide full confidence in the interoperability of the PBX and Call2Teams.

Create at least 2 users on the PBX and connect to Call2Teams. They are called User A and User B. They should have active DiD numbers.

7.1 Single calls

7.1.1 Single Incoming call 1

	Description	Expected Result
Step 1	From an external number, dial the User A PBX number	Teams rings for User A The caller-id matches the external number that originated the call
Step 2	Answer the call in Teams	2-way audio is established
Step 3	Speak for 45 seconds	The call continues to work
Step 4	Place the call on hold in Teams	The caller hears hold music

Step 5	Resume the call in Teams	2-way audio is re-established
Step 6	Blind transfer the call to another external number	The external number rings The caller-id is as expected (either User A's DiD or the external number that originated the call, depending on how the PBX is configured)
Step 7	Answer the call	2-way audio is established between the two external devices. Teams hangs-up.

7.1.2 Single Incoming Call 2

	Description	**Expected Result**
Step 1	From an external number, dial the User A PBX number	Teams rings for User A
Step 2	Answer the call in Teams	2-way audio is established
Step 3	Blind transfer the call to User B's extension number	Teams rings for User B
Step 4	Answer the call	2-way audio is established between the two external devices. External Caller hangs up.

7.1.3 Single Outgoing Call 1

	Description	**Expected Result**
Step 1	From User A's Teams client, dial an external number	The external number rings.
Step 2	Answer the call	2-way audio is established
Step 3	Speak for 45 seconds	The call continues to work
Step 4	Place the call on hold in Teams	The caller hears hold music
Step 5	Resume the call in Teams	2-way audio is re-established
Step 6	Blind transfer the call to another external number	The external number rings
Step 7	Answer the call	2-way audio is established between the two external devices. Teams hangs-up.

7.1.4 Single Outgoing call 2

	Description	**Expected Result**
Step 1	From User A's Teams client, dial User B's extension	User B's Teams client rings
Step 2	Answer the call	2-way audio is established

Step 3	Speak for 45 seconds	The call continues to work User A hangs up.
Step 6	Blind transfer the call to another external number	The external number rings

7.2 Multiple calls

This step confirms that the PBX can handle multiple users registering from the same IP and port.

7.2.1 Two incoming calls on different Users

	Description	**Expected Result**
Step 1	From an external number, dial User A's DiD	Teams rings for User A
Step 2	From an external number, dial User B's DiD	Teams rings for User B
Step 3	Both Teams users answer	2-way audio is established on both calls
Step 4	Teams hangs-up the first call	Both parties see the call hang up
Step 5	External hangs-up the second call	Both parties see the call hang up

7.2.2 Two outgoing calls on different users

	Description	**Expected Result**
Step 1	From User A's Teams client, dial an external number	The external number rings
Step 2	From User B's Teams client, dial an external number	The external number rings
Step 3	Both external numbers answer	2-way audio is established on both calls
Step 4	Teams hangs-up the first call	Both parties see the call hang up
Step 5	External hangs-up the second call	Both parties see the call hang up