



Skype Connect™

Troubleshooting Guide

Version 4.0

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1.0 Support resources

Skype provides a range of self-help resources for both technical and non-technical issues. All support resources are easily accessed via support.skype.com. Here you will find Skype Connect links to:

- User guides and documentation
- FAQs - support.skype.com/category/SKYPE_CONNECT
- An expert community forum
- The option to raise a Skype Customer Support ticket

In addition, Skype's dedicated Business Support Team is available to answer Skype Manager™, Skype Connect™, account management and billing queries by email or chat. Just go to support.skype.com, click on **Get more help** and follow the on-screen instructions.

You can also check the status of Skype at any time by visiting the Heartbeat page and reviewing the **Skype Connect** heartbeat. Please see heartbeat.skype.com.

2.0 Troubleshooting quick reference

Symptom	Cause	Fault Replacement	Diagnostic Points	Potential Solutions
Mid call dropping	<ul style="list-style-type: none"> The RTP stream is disconnecting 	<ul style="list-style-type: none"> Connectivity 	<ul style="list-style-type: none"> Check trace at the SIP-enabled PBX 	<ul style="list-style-type: none"> Check internet connectivity
Mid call dropping	<ul style="list-style-type: none"> The Skype Credit runs out mid call 	<ul style="list-style-type: none"> Skype Manager 	<ul style="list-style-type: none"> Check the SIP Profile's Skype Credit 	<ul style="list-style-type: none"> Top up the SIP Profile's Skype Credit
Calls from the Online Number are not reaching SIP-enabled PBX	<ul style="list-style-type: none"> The Online Number has expired The Skype Connect channel subscription has run out The SIP-enabled PBX is not registered The Online Number has failed 	<ul style="list-style-type: none"> Skype Manager SIP-enabled PBX Skype 	<ul style="list-style-type: none"> Check the SIP Profile's Skype Connect channel subscriptions Check the Skype Manager for the registration status and SIP-enabled PBX Check Skype 	<ul style="list-style-type: none"> Extend the Online Number Activate the Skype Connect channel subscription Re-register the SIP-enabled PBX Wait for Skype
Calls to landline or mobile numbers are failing	<ul style="list-style-type: none"> Skype Credit has run out Internet connection problems Skype Connect problems 	<ul style="list-style-type: none"> Internet connection Skype Manager Skype 	<ul style="list-style-type: none"> Check the internet connection Check the SIP Profile's Skype Credit 	<ul style="list-style-type: none"> Top up the SIP Profile's Skype Credit Wait for Skype
The audio quality is poor	<ul style="list-style-type: none"> Problems with your local area network Internet connection problems Skype Connect problems 	<ul style="list-style-type: none"> LAN ISP Skype Connect 	<ul style="list-style-type: none"> Check for sufficient bandwidth Check the Skype Connect heartbeat 	<ul style="list-style-type: none"> Buy a Quality of Service router Upgrade your bandwidth Wait for Skype Connect
Voice problems	<ul style="list-style-type: none"> The RTP stream is blocked 	<ul style="list-style-type: none"> NAT Firewall SBC settings 	<ul style="list-style-type: none"> Check the NAT/firewall Check the SIP-enabled PBX configuration 	<ul style="list-style-type: none"> Buy a SIP diagnostic utility program
The SIP-enabled PBX features are not working, such as Call Hold, Call Transfer and Conferencing	<ul style="list-style-type: none"> SIP-enabled PBX problems Skype Connect problems 	<ul style="list-style-type: none"> SIP-enabled PBX 	<ul style="list-style-type: none"> Check the SIP-enabled PBX configuration Check the Skype Connect certified SIP-enabled PBX 	<ul style="list-style-type: none"> Buy a Skype Connect certified SIP-enabled PBX
DTMF is not working	<ul style="list-style-type: none"> SIP-enabled PBX problems Skype Connect problems 	<ul style="list-style-type: none"> SIP-enabled PBX Skype Connect 	<ul style="list-style-type: none"> Check the SIP-enabled PBX configuration (make sure it is set for RFC2833) Check Skype Connect 	<ul style="list-style-type: none"> Configure the SIP-enabled PBX to use RFC2833

3.0 Common issues checklist

If you're unsure how to perform specific actions within Skype Manager, please see the [Skype Connect User Guide](#).

3.1 You can't make outgoing calls to landline and mobile numbers

Check	Solution
1. Have you run out of Skype Credit?	<ul style="list-style-type: none">• Sign in to your Skype Manager at manager.skype.com and allocate more Skype Credit to the SIP Profile. We recommend that you enable Auto-recharge on your SIP Profile as well.
2. Is your SIP-enabled PBX's internet connection 'live'?	<ul style="list-style-type: none">• If possible, test your internet connection by browsing to a web page. If the connection isn't working, contact your Internet Service Provider.• Check your SIP-enabled PBX's registration status in Skype Manager.
3. Is your SIP-enabled PBX connected to Skype Connect?	<ul style="list-style-type: none">• Check your SIP-enabled PBX has completed its SIP registration process with Skype Connect.• Your Skype Manager and SIP Profile usernames and passwords are different. Check that you are using the same SIP Profile authentication credentials shown in Skype Manager and check that the correct SIP Profile username and password are in your SIP-enabled PBX's configuration.• Reset the credentials in Skype Manager and enter the new credentials into your SIP-enabled PBX.
4. Is there an outage or planned maintenance on Skype Connect?	<ul style="list-style-type: none">• Check whether there is an outage by visiting heartbeat.skype.com and reviewing the Skype Connect heartbeat.

3.2 You have Quality of Service (QoS) issues on your corporate network

Check	Solution
<p>1. Is voice data prioritized over other data on your corporate network?</p>	<ul style="list-style-type: none">• If you have a Quality of Service-based router on your corporate network, check its configuration to see if voice data is prioritized (ports 8000-80xx - where “xx” is the number of channels you have, multiplied by two).• If you don't have a Quality of Service-based router on your corporate network, we recommend that you buy one, especially if you are using the same internet connection for your voice and other data.

3.3 You have SIP-enabled PBX DTMF issues

Check	Solution
<p>1. Are you using a Skype Connect certified SIP-enabled PBX and RFC 2833 DTMF?</p> <p>If you're not using a Skype Connect certified SIP-enabled PBX, you may experience PBX and DTMF issues.</p>	<ul style="list-style-type: none">• Check that your SIP-enabled PBX is Skype Connect certified. A list of Skype Connect certified SIP-enabled PBXs is available at skype.com/go/sip/vendors.• Check that your SIP-enabled PBX is configured to use RFC 2833 (sometimes described as 'out of band') DTMF with Skype Connect.• We recommend that your SIP-enabled PBX uses RTP Payload Type 101 for the SIP message's telephone-event.

3.4 You don't know if Skype Connect can do what you need it to do

Check	Solution
<p>1. Have you checked the Skype Connect support resources?</p>	<ul style="list-style-type: none">• Check support.skype.com to see if a particular feature is present. Alternatively, please see skype.com/go/sip.

4.0 Common issues in detail

This chapter examines the following types of problems in more detail, to help you resolve them:

1. Call failure issues

Problems such as:

- 'I can't make calls to landline or mobile numbers'
- 'I'm not receiving incoming calls in my SIP-enabled PBX'
- 'Calls keep dropping mid-call'

2. Voice quality issues

Problems such as:

- 'Inaudible audio'
- 'Audio playback is chopping in and out'
- 'Voice playback is too fast or too slow'
- 'Audio playback sounds like a robot'

3. Voice issues

Problems such as:

- 'I can't hear the caller'
- 'The person I called can't hear me'

4. PBX functionality issues

Problems with:

- Call Hold
- Call Transfer
- Music on Hold
- Conference Calling
- Hunt Groups
- Your internal network routing

5. DTMF (dial pad tones) issues

Problems such as:

- When you connect to an auto attendant and are asked to enter a keypad number you get no response – “your number has not been recognized, please try again”.
- If asked to enter a PIN via your telephone's key pad and there is no response.

4.1 Call failure issues

Issues with your Skype Credit, SIP-enabled PBX setup or planned maintenance may affect your ability to make or receive calls.

4.1.1 You cannot call mobile or landline numbers when using Skype Connect

Check	Solution
<p>1. Have you sufficient Skype Credit allocated to your SIP Profile?</p> <p>Skype Connect is a Pay As You Go product. You will be unable to make any outgoing calls if you:</p> <ul style="list-style-type: none">• Have no Skype Credit allocated to your SIP Profile.• Have not paid the monthly recurring fee for your Skype Connect channel subscription.	<ul style="list-style-type: none">• Sign in to your Skype Manager at manager.skype.com and allocate more Skype Credit to the SIP Profile.• Skype Connect channel subscriptions for a SIP Profile are paid from Skype Manager's credit balance, not from the Skype Credit allocated to the SIP Profile. Add more Skype Credit to Skype Manager if there is not enough available to pay for the Skype Connect channel subscriptions.• Consider using Auto-recharge for your SIP Profiles to avoid a recurrence of this issue. With Auto-recharge, Skype Credit is automatically allocated from Skype Manager to your SIP Profiles to pay for calls.
<p>2. Is the connection your SIP-enabled PBX uses to connect to the internet 'live'?</p> <p>If your SIP-enabled PBX loses its connection to the internet, it won't be able to connect to Skype, resulting in the complete loss of calling. Any internet connection may fail for short or long periods, on an intermittent or rare basis.</p> <p>Loss of connectivity to the internet will produce the following kinds of errors:</p> <ul style="list-style-type: none">• RTP in stream failure.• Registration Request Failure.• Registration Request Timeout.	<ul style="list-style-type: none">• If possible, test the internet connection by browsing to a web page. If the connection isn't working, contact your ISP.• Run our approved test tool at tools.voxygen.co.uk. This will tell you whether your corporate network can connect to Skype.• If your internet connection is working, proceed to the next check.

<p>3. Is your SIP-enabled PBX connected to Skype?</p> <p>If your internet connection is working, you may have lost calling ability if your SIP-enabled PBX is unable to connect to Skype. This is normally due to a problem with your SIP credentials, firewall, or router.</p> <p>The following kinds of errors will be produced:</p> <ul style="list-style-type: none"> • Registration Request Timeout • Registration Request Failure • Invite failed 	<ul style="list-style-type: none"> • Check whether your Skype Connect authentication credentials have been reset, or if your SIP Profile has been deleted from Skype Manager. • Check whether anything on your corporate network has changed, such as the replacement of one of your routers or the addition of a NAT, router, firewall or other device to your communications system. If so, run the test tools and configure the new device accordingly, ensuring that access to Skype Connect is available (UDP ports 5060, 8000-800xx where “xx” is number of channels you have, multiplied by two). • Ensure the authentication credentials are correct. Sign in to your Skype Manager at manager.skype.com and check the SIP Profile’s credentials against those in the SIP-enabled PBX. • Check if the registration has timed out on your SIP-enabled PBX and the PBX is not configured to auto-retry. • Check the registration status in Skype Manager. • Check whether there is an outage by visiting heartbeat.skype.com and reviewing the Skype Connect heartbeat. • Reset the SIP Profile's credentials in Skype Manager. • Contact your PBX manufacturer for more help.
<p>4. Is there an outage or planned maintenance on Skype Connect?</p>	<ul style="list-style-type: none"> • Check whether there is an outage by visiting heartbeat.skype.com and reviewing the Skype Connect heartbeat.

4.1.2 You cannot receive incoming calls

It is possible for a situation to arise in which other functionality on your SIP Profile is working, but you are not receiving incoming calls from Skype, landlines or mobiles.

Check	Solution
<p>1. Has an Online Number for that SIP Profile expired?</p> <p>Online Numbers can be linked with a SIP Profile either directly by adding it to the SIP Profile or by association with a business Skype account (business Skype Name) that has been added to the SIP Profile. Online Numbers can receive calls to the same SIP Profile.</p> <p>If calls to one, or some, of those Online Numbers are not working, it may be that the Online Numbers are expiring without being renewed by the Skype Manager administrator (Online Numbers are created with an expiry date).</p>	<ul style="list-style-type: none"> • Sign in to your Skype Manager at manager.skype.com and check whether the Online Numbers affected are expiring. <p>Important: Online Numbers cannot be automatically extended and must be consciously extended by your Skype Manager administrator.</p>

4.1.3 Your calls keep dropping mid-call

The fact that you can start a Skype Connect call shows that:

- Your SIP-enabled PBX has (possibly intermittent) connectivity to the internet.
- The SIP-enabled PBX's SIP Profile credentials are correct.
- You have Skype Credit.

Check	Solution
<p>1. Is your internet connection losing connectivity on an intermittent basis?</p>	<ul style="list-style-type: none"> • If you are using a dedicated or shared internet connection for voice, contact your ISP regarding loss of connectivity on an intermittent basis. Your ISP can track such occurrences.
<p>2. Is there a problem with your SIP-enabled PBX?</p>	<ul style="list-style-type: none"> • If you use multiple communication suppliers on one SIP-enabled PBX, check whether you are experiencing call drops using their services. • Contact your PBX vendor or manufacturer for further assistance.

4.2 Voice quality issues

Internet voice quality is a subjective issue (as opposed to something quantifiable, such as ‘we can’t make any calls’) and can be quite challenging to troubleshoot post-implementation. We have aimed to provide sufficient detail in this section to support you in relation to Skype Connect.

Calls made using Skype Connect travel over the public internet and your own corporate network. Skype does not control your corporate network or the public internet, therefore Skype cannot control the audio quality of experience and such experience cannot be guaranteed.

4.2.1 Manifestations of poor internet voice quality

Poor internet voice quality is typically exhibited by choppy, clipped, distorted or cartoon-sounding audio.

There are three main causes of poor Quality of Experience with internet voice:

- The performance and suitability of the internet connection that the SIP-enabled PBX uses.
- Voice data competing for bandwidth with other forms of data on your corporate network or internet connection does not have priority.
- Internet congestion, i.e. issues experienced by your internet service provider.

4.2.2 Understanding the impact of your internet connection on Quality of Experience

Internet voice is a real-time communication method. The data packets that make up voice conversations need to arrive to and from your SIP-enabled PBX in a timely manner to support natural conversations. If packets are delayed, arrive out of sequence, or are lost completely, then the quality of voice audio suffers.

The parameters that measure this on an internet connection are known as latency, jitter and packet loss. Although quality remains a subjective term, the lower these parameters measure, the better the Quality of Experience is likely to be.

Issues with latency, jitter or packet loss can occur individually or in any combination on an internet connection.

1. Latency

- Latency is a consistent delay in the reception and transmission of data.
- Latency doesn’t matter for data packets for web browsing or email because, if they arrive even a few milliseconds late, you are unlikely to notice. But it does matter with voice.
- It is measured in milliseconds. A delay as small as half a second (500ms) is noticeable.

- The acceptable range for Skype Connect is 250ms route trip delay.
- High latency will result in a noticeable delay between you talking and the other person receiving that audio.
- Additionally, in some circumstances a high latency can cause echo.

2. Jitter

- Jitter is a variable delay in the reception and transmission of data packets.
- It is measured in percentage terms.
- A high level of jitter produces a “Donald Duck” or “Dalek” effect.
- You can reconfigure your PBX with larger Jitter Buffers to help combat this problem.

3. Packet loss

- Packet loss is, as the name suggests, the loss of data packets in the transmission of data over the internet or your corporate network.
- The sound of a speaker’s voice is deconstructed into data packets at one end, transmitted and reconstructed at the other end. The deconstruction and reconstruction of voice data packets is handled by a CODEC.
- If data packets are lost during the transmission phase voice quality will suffer because part of the original audio signal has, literally, been lost.
- It is measured in percentage terms.
- The acceptable range for Skype Connect is 2ms.
- High levels of packet loss produces 'holes' or 'drop outs' in the audio.

These measures relate directly to the performance of your internet connection and of your local corporate network. If you are experiencing Quality of Experience issues, you may find that performance varies during the day, or that performance is particularly bad at specific times (also see voice data priority). Skype does not control your corporate network or the public internet, therefore Skype cannot control the audio quality of experience and such experience cannot be guaranteed.

Check	Solution
1. Have you checked the Quality of Service at different times of the day?	<ul style="list-style-type: none"> • We recommend that you run four tests at different times of the day to establish whether there are any patterns to your issues.
2. Are you using the G.711 codec?	<ul style="list-style-type: none"> • See if your SIP-enabled PBX supports G.729. G.729 is both less bandwidth intensive, and has better error correction than G.711, and typically performs better in poor corporate network conditions.

3. Is your internet connection providing an acceptable performance?	<ul style="list-style-type: none"> If your internet connection does not provide acceptable performance, you may need to consider either upgrading or changing ISP.
4. Do you use the same internet connection for voice and data?	<ul style="list-style-type: none"> Consider switching to a dedicated connection for voice traffic.

4.2.3 Voice data prioritization on your internet connection and your corporate network

Just as the arrival of data packets can be delayed due to jitter or latency on your corporate network or internet connection, a noticeable delay can also be caused by voice data having to compete for bandwidth with other data, such as web browsing.

In simple terms, packets must wait to arrive or to be sent. Your experience of web browsing is not affected if packets are queuing for a few milliseconds. However, you will notice if voice data packets are regularly ‘queuing’.

The idea of competing data applies primarily to implementations where voice and other data share the same internet connection. However, these issues can also occasionally occur on dedicated voice connections at a corporate LAN level, particularly where the corporate LAN network traffic demands are large, for example, in large companies.

Check	Solution
1. Do you use a Quality of Service router?	<ul style="list-style-type: none"> If you don't currently use a Quality of Service router, buy a specialized SIP Quality of Service-based router for your corporate network.
2. Have you configured your Quality of Service router appropriately?	<ul style="list-style-type: none"> Enable voice data prioritization in the configuration of your router. For more information on how to do this, please see your router manufacturer's support documentation.

4.2.4 Internet congestion

Internet congestion can cause Quality of Experience issues with calls to a particular part of the world. This can happen even if you have an excellent internet connection because your ISP does not control the end-to-end connection. To complete the call, data packets may need to travel over other carriers' infrastructure, which may experience issues of their own.

Check	Solution
1. Are you using the G.711 codec?	<ul style="list-style-type: none"> If you are using the G.711 codec, see if your SIP-enabled PBX supports G.729. G.729 is both less bandwidth intensive, and has better error correction than G.711, and typically performs better in poor corporate network conditions.

4.3 Voice issues

Voice problems are typically caused by voice packets being blocked on the associated incoming or outgoing port of your router or NAT device or by your ISP.

Check	Solution
1. Are you using a NAT, router or firewall device in between your corporate LAN and internet connection?	<ul style="list-style-type: none"> Ensure that it does not restrict the use of the following ports: UDP 5060, 8000-80xx (where "xx" is the number of channels you require multiplied by two). For more help, please see your device's support documentation or contact the device manufacturer for more help.

4.4 PBX functionality issues

Problems with any of the following will be caused by your SIP-enabled PBX:

- Hunt Groups
- Hold
- Transfer
- Music on Hold
- Internal network routing

If you are experiencing any of these issues, contact your PBX manufacturer for help specific to your hardware and corporate network.

4.5 TLS/sRTP issues

Check	Solution
1. Have you configured your firewall for Secure Connect (TLS/sRTP)?	<ul style="list-style-type: none">Your firewall rules need to allow:<ul style="list-style-type: none">Outbound DNS SRV requests by Skype to the domain sip.skype.comReceipt of DNS SRV records by SkypeOutbound TCP/IP connections from your public interface to the IP addresses provided in the DNS SRV record from SkypeReceipt of incoming TCP/IP from Skype Connect
2. Have you configured your device for TLS?	<ul style="list-style-type: none">Ensure that all of your devices using Skype are configured for TLS and sRTP. Assign TLS to port 5061. sRTP will be registered in the SIP session using SDES. Configure your devices to perform DNS SRV requests to sip.skype.com.
3. Have you configured your device (PBX or Gateway) to use TLS and sRTP (optional)?	<ul style="list-style-type: none">Some devices need the CA (Verisign) signed certificate pre-installed in the SIP User Agent (UA) configuration. Please get this certificate from your Skype Connect technical contract.
4. Have you configured your SIP Profile using the Registration method (Username/Password option)?	<ul style="list-style-type: none">SIP Profiles using IP Authentication do not support the TLS/sRTP feature in this release. It will be supported in a future release.

4.6 DTMF (dial pad tones) issues

Check	Solution
1. Are you using RFC 2833 (out of band) DTMF?	<ul style="list-style-type: none">Ensure your SIP-enabled PBX is configured to use RFC 2833 DTMF and not in-band DTMF.

2. Are you using uncertified equipment?

- Ensure you are using Skype Connect certified equipment. Due to the complexity of SIP equipment, Skype has a Skype Connect certification programme to confirm that certain equipment is interoperable with Skype Connect. Use of uncertified equipment may lead to problems with functions such as DTMF. For more information on Skype Connect certified SIP-enabled PBXs and gateways, please see skype.com/go/sip.vendors.

Access to a broadband internet connection is required. Skype is not a replacement for traditional telephone services and cannot be used for emergency calling. Skype Connect is meant to complement existing traditional telephone services used with a corporate SIP-enabled PBX, it is not a stand-alone solution. Skype Connect users need to ensure all calls to emergency services are terminated through traditional fixed line telephone services, connected to the local exchange, or through other emergency calling capable telephone services.

5.0 Skype Connect response codes

5.1 Response code classification

The following table provides an overview of how the Skype Connect response codes are classified according to number range. These response types are defined by RFC3261:

ietf.org/rfc/rfc3261.txt

Number range	Description
1xx	Information and provisional response to a SIP request.
2xx	Success response to a SIP request.
3xx	Redirect response to a SIP request.
4xx	Client error response to a SIP request.
5xx	Server error response to a SIP request.
6xx	Global response to a SIP request.

5.2 Response codes

The following tables provide a list of Skype Connect response codes, categorized by response type.

5.2.1 Authentication response codes

Request	Response	Remedy
The REGISTER request does not contain the following in the REGISTER headers: <ul style="list-style-type: none">• A valid SIP username• Skype domain "sip.skype.com"	401	Ensure that the REGISTER header is configured with: <ul style="list-style-type: none">• A valid SIP username• Skype domain "sip.skype.com"

The REGISTER authorisation details in response to a "401 Unauthorized" header do not match the SIP username/password and domain.	403	Ensure that the SIP username and password for the SIP Profile have been entered correctly into your SIP-enabled PBX.
The SIP-enabled PBX attempts to send another REGISTER request before the previous REGISTER timer expires.	503	Ensure SIP-enabled PBX re-registers using the expiry timer specified by Skype Connect's 200OK response code.
IP Authentication INVITE request with an invalid SIP username in the User portion of the From header of the SIP URI.	403	Ensure SIP username is in the User portion of the From header of the SIP URI.

5.2.2 SIP Account credit and subscriptions response codes

Request	Response	Remedy
The SIP Profile does not have enough Skype Credit to make an outbound call.	403	Allocate more Skype Credit to the SIP Profile and set up Auto-recharge on the SIP Profile to ensure you do not run out of Skype Credit again.
The SIP Profile has no available Skype Connect channel for making an outbound call.	403	Buy additional Skype Connect channels for your SIP Profile.
The SIP Profile is inactive due to expiry of the Skype Connect channel subscription when making an outbound call.	480	Ensure Skype Manager has sufficient Skype Credit to renew your Skype Connect channel subscription.

5.2.3 Calling condition response codes

Request	Response	Remedy
The called party is busy.	486	Call again.
The called party rings with no answer.	408	Call again.
The called party is out of service.	502	Check with the called party.
The called party is out of area or coverage.	200OK	Call again.

The dialled number is not reachable with Skype Connect.	404	Dialled number is not supported by Skype. For more information on the numbers supported, please see skype.com/business/sip/pricing
The dialled destination is not reachable from user's calling country.	403	Dialled number is not supported by Skype. For more information on the numbers supported, please see skype.com/business/sip/pricing
The dialled number is blocked by Skype.	403	Dialled number is not supported by Skype. For more information on the numbers supported, please see skype.com/business/sip/pricing

5.2.4 SIP request response codes

Request	Response	Remedy
The INVITE request does not contain SIP user and "sip.skype.com" in the From header. For example: 9905xxxxxxx@sip.skype.com (DIGEST PROFILES).	403	Ensure that your SIP-enabled PBX has been correctly configured to use the SIP Profile credentials. All Skype Connect certified SIP-enabled PBX manufacturers will have instructions on how to configure your PBX with Skype Connect. For more information on Skype Connect certified SIP-enabled PBX manufacturers, please see skype.com/go/sip/vendors

<p>The INVITE request does not contain "sip.skype.com" in the Invite Request Line header. For example:</p> <p>diallednumber@sip.skype.com</p>	403	<p>Ensure that your SIP-enabled PBX has been correctly configured to use the SIP Profile credentials.</p> <p>All Skype Connect certified PBX manufacturers will have instructions on how to configure your PBX with Skype Connect.</p> <p>For more information on Skype Connect certified SIP-enabled PBX manufacturers, please see skype.com/go/sip.vendors</p>
<p>The INVITE request uses a domestic and not International number format in the Request Line.</p>	404	<p>Ensure that the number used (either by a user or your SIP-enabled PBX) is in the full E.164 international telephone number format.</p>
<p>The INVITE request is challenged with a "407 Proxy Authentication required", SIP PBX does not respond with Authorization header in INVITE.</p>	403	<p>Ensure that your SIP-enabled PBX is configured to respond with Authorisation Digest information.</p>
<p>The INVITE request contains a Codec in SDP that is not supported.</p>	488	<p>Skype supports G.711 and G.729.</p>
<p>The method requested is not supported.</p>	405	<p>Ensure that your SIP-enabled PBX has been correctly configured for use with Skype Connect.</p> <p>All Skype Connect certified PBX manufacturers will have instructions on how to configure your PBX with Skype Connect.</p> <p>For more information on Skype Connect certified SIP-enabled PBX manufacturers, please see skype.com/go/sip.vendors</p>
<p>The INVITE request contains an invalid Domain in Request line (valid domain is "sip.skype.com").</p>	403	<p>Ensure the domain in your SIP-enabled PBX configuration is "sip.skype.com".</p>

The INVITE request contains Max-Forwards=0	487	<p>Ensure that your SIP-enabled PBX has been correctly configured for use with Skype Connect.</p> <p>All Skype Connect certified PBX manufacturers will have instructions on how to configure your PBX with Skype Connect.</p> <p>For more information on Skype Connect certified SIP-enabled PBX manufacturers, please see skype.com/go/sip.vendors</p>
The INVITE request line contains an incomplete number.	404	Ensure SIP-enabled PBX uses E.164 number format.
Skype cannot accept any more requests.	501	<p>Check whether there is an outage by visiting heartbeat.skype.com.</p> <p>If no outage is listed, contact Skype Customer Service for details.</p>
Skype fails to parse Request.	501	<p>Check whether there is an outage by visiting heartbeat.skype.com.</p> <p>If no outage is listed, contact Skype Customer Service for details.</p>
Skype hangs up in error.	503	<p>Check whether there is an outage by visiting heartbeat.skype.com.</p> <p>If no outage is listed, contact Skype Customer Service for details.</p>

