skype™

Skype Connect™
Requirements Guide

Version 4.0

Thinking about implementing Skype Connect? Read this guide first.

Skype Connect™ provides connectivity between your business and the Skype community. By adding Skype Connect to your existing SIP-enabled PBX, your business can save on communication costs with little or no additional upgrades required.

This guide aims to help you understand the technical requirements of Skype Connect and to determine whether you need to make any equipment or infrastructure changes within your business to be able to use it.

For more information on setting up and using Skype Connect, please see the **Skype Connect User Guide**.

Contents

1.0 What you need to use Skype Connect

Before using Skype Connect, you need the following:

- **Skype Manager™**

  Skype's business products, including Skype Connect, can be conveniently managed from Skype Manager where you can create SIP Profiles for your Skype Connect service. If you require inbound calling, you can add business Skype accounts and Online Numbers. If you require outbound calling, you can allocate Skype Credit. You can also set up Caller ID and view reports, all in one place.

  To find out more about Skype Manager, please see the **Skype Manager User Guide**. To find out more about using Skype Connect within Skype Manager, please see the **Skype Connect User Guide**.

- **A Skype Connect certified SIP-enabled PBX or gateway**

  We recommend that you use a Skype Connect certified SIP-enabled PBX as we cannot guarantee all features will operate correctly in non-certified PBXs when used with Skype Connect. Alternatively, if you do not have a SIP-enabled PBX, a Skype Connect certified gateway can be installed directly on your outside line so you don't need to re-configure your PBX.

  For the current list of Skype Connect certified SIP-enabled PBX and gateway vendors, please see **skype.com/go/business.sip.pbx.vendors**.

- **An internet connection with sufficient bandwidth to handle your voice traffic**

  This can be either a shared connection or dedicated connection, depending on your business requirements.

- **An internet connection of suitable quality for internet voice communications**

  Your internet connection should provide less than 200ms Round Trip Delay between your business LAN and sip.skype.com.

  You can check the suitability of your internet connection by checking the Round Trip Delay (RTD) from your business network (LAN) to Skype Connect. To do this, try sending a ping to sip.skype.com and note the average time shown in the ping results.

- **Publicly reachable IP Address**

  If your SIP-enabled PBX is located on a private network behind a NAT firewall or router, you can still use Skype Connect. To do this, you should select the SIP registration method of authentication when creating the SIP Profile.

  Alternatively, you can use a static IP address if your SIP PBX does not support registrations or you do not wish to use the registration method of authentication. To do this, you should select the IP Authentication method when creating the SIP Profile.

- **A means to pre-pay for Skype products**

   As a pre-pay offering, you will need to be able to buy and Auto-recharge Skype Credit. A number of payment methods are available, including:

   - PayPal account

   - Credit or debit card

   - Bank wire transfer

   A full list of the payment methods that are available can be found at **skype.com/go/ paymentmethods**.

   When you sign up for Skype Manager, you are given a purchase limit that determines the maximum amount of Skype Credit you can purchase in one go. The range of payment methods available to you is also limited. To access a wider range of payment methods, or to increase your purchase limit, you will need to get your company verified. For more information on how to get your company verified, please see the **Skype Manager User Guide**.

- **Traditional fixed line telephone connectivity**

   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.

2.0 Skype Connect features and restrictions

2.1 Features

- Skype Connect supports the SIP registration and IP Authentication method, depending on what your SIP-enabled PBX supports and your company's security policy.

- The maximum number of simultaneous channels you can use for Skype Connect is restricted to 300 per SIP Profile. However, you can have as many SIP Profiles as you like (if your PBX supports it).

- All outbound calls are charged at Skype Connect's standard rates and call costs are debited from your SIP Profile's credit balance. However, if you call the US a lot, you may save on your calling by adding US calling minute bundles.

  Each bundle includes a monthly allowance of 5,000 minutes to call landlines and mobiles in the US. If your minutes run out before the next renewal date, calls to the US will be charged from your SIP Profile's available Skype Credit balance at Skype Connect's standard call rates.

  Important: The number of US minute bundles you add to your SIP Profile cannot exceed the number of channels you have added to the SIP Profile.

- You can add an unlimited number of business Skype accounts to a SIP Profile to receive calls from Skype users worldwide. Skype calls to the Skype Name of that business Skype account will be directed to the SIP Profile and the associated SIP-enabled PBX.

  A Skype button using the Skype Name of that business account can be created and placed on, for example, your company's website, so that customers with Skype can call your SIP Profile using Skype for free. Your company is therefore immediately connected to millions of registered Skype users worldwide.

  In addition, if the business Skype account has any Online Numbers allocated to it, then calls to those Online Numbers will also be received in your SIP-enabled PBX. This potentially offers you the option to create DID (Direct Inward Dial) functionality, depending on the communications equipment you are using.

  Important: A business Skype account may only be added to one SIP Profile. If a business Skype account is assigned to a SIP Profile it cannot be used to sign in to Skype on your computer or any other device.

  We also strongly recommend that you do not add your Skype Manager administrator's account to a SIP Profile under any circumstances as this would prevent the administrator from being able to sign in to Skype Manager.

- Adding an Online Number to a SIP Profile allows calls to that Online Number to be received in your SIP-enabled PBX. Depending on your PBX, you may also be able to

configure it to direct calls to that Online Number to your SIP-enabled PBX extensions and resources such as extension numbers, Hunt Groups, Auto Attendants or Contact Center.

- You can use both business Skype accounts and Online Numbers simultaneously on a single SIP Profile if you wish. In other words, you may have multiple business Skype accounts and Online Numbers added to one SIP Profile at the same time.

- Caller ID is supported for certain Online Numbers provided by Skype. Landline numbers can also be used as Caller ID so long as the numbers are owned and controlled by you and your company identity has been verified by Skype. If you don't set up Caller ID, "Unknown number" will be displayed when you make an outbound call using Skype Connect.

    Important: Landline numbers and/or Online Numbers from certain countries may not be able to be used as Caller ID.

- Skype Connect will send you both the Online Number and the caller's Caller ID in the SIP messages so you can easily map incoming calls within your SIP-enabled PBX.

- Skype Connect includes industry standard support for Transport Layer Security (TLS) and Secure RTP (sRTP) which encrypts SIP messages and RTP media transported between your SIP-enabled PBX and Skype Connect. This helps to protect your calls and provides a higher level of security for your business communications.

    TLS and sRTP are automatically enabled so no configuration or provisioning is required (where supported by your SIP-enabled PBX).

2.2 Restrictions

- Skype Connect does not support special service numbers, e.g. 411, or operator services and informational services.

- Skype Connect does not support faxing.

- Skype Connect is not guaranteed to work with credit card machines, franking (stamping) machines and alarm systems or other services which use a regular phone line with a modem connection.

- Video calls using Skype Connect are not supported.

- Call forwarding to the Skype Names of business Skype accounts that have been added to a SIP Profile is not supported.

3.0 System requirements

Skype Connect has been designed to work with existing SIP-enabled PBX systems. Whether your internet connection is Analog, Digital (T1/ISDN) or SIP, Skype Connect can be integrated with your communications infrastructure

Here are the system requirements of Skype Connect:

Requirement	Description
Protocol	SIP (RFC3261)
Far End NAT traversal	NAP/T
Privacy extensions	RFC3325
IP transport	UDP and TCP/IP
Authentication	SIP DIGEST or IP Authentication These are requested for both REGISTER and INVITE
Registration	Automatic NAT detection and change of registration timers
Channels	Maximum of 300 simultaneous calls per SIP Profile
Codecs	G.729 and G.711 ulaw/ALaw
DTMF method	RFC2833
Call type support	Skype Connect supports the following third party call control features when used in conjunction with a Skype Connect certified SIP-enabled PBX: • Mid call codec change • Re-INVITE for various third party call control features including Call Hold, Call Transfer, Park, Call Divert and other types of SIP-enabled PBX call types • RFC3325 for P-Asserted-ID, Remote-Party-ID and Privacy:id based calls
Number format	E.164 (country code and national number), international number format for all calls
Fax	Skype does not support Fax or T.38
Emergency calls	Skype does not support calls to emergency services

4.0 Technical considerations

In this chapter, we highlight the major technical considerations for implementing Skype Connect, along with some recommendations. We will cover:

- Quality of Service (QoS)
- Voice
- Channel capacity
- Impact on other data transfer rates
- Transport Layer Security (TLS) and Secure RTP (sRTP)
- Geographic redundancy
- LAN connectivity

4.1 Avoiding Quality of Service issues

Quality of Service (QoS) is a major consideration for any new or existing communications implementation. However, getting a few things right will help avoid quality problems.

Skype Connect is a 'real time' communications product and is therefore time-sensitive. It has a particular set of requirements which are not the same as non-real time services, such as web browsing and email. Providing good Quality of Service depends on having as little delay in your corporate network as possible, including your internet connection. Voice data needs to be shared at speeds and a quality that enables normal conversations.

Calls made using Skype Connect will travel over the public internet and your own corporate network. As Skype does not control your corporate network or the public internet, Quality of Service cannot be guaranteed.

4.1.1 Causes of Quality of Service issues

The primary causes of quality issues with internet voice communications are:

- High latency, jitter or packet loss on your internet connection or your local corporate network.
- On a shared internet connection, voice traffic is given insufficient priority over time-insensitive data on your corporate network.
- Your internet connection is oversubscribed and is asymmetric. For example, ADSL lines support more downstream traffic than upstream, and a Skype Connect call requires an equal amount of both up and down stream capacity.

4.1.2 Recommendations for avoiding Quality of Service issues

We recommend the following to help to avoid Quality of Service issues:

- **Ensure you have an internet connection which is suitable for communicating over the internet**

  As a guide, an internet connection to be used for communicating over the internet should be of sufficient quality to provide:

  - Low packet loss of less than 0.2%
  - Latency from your business IT network to sip.skype.com should be less than 200 milliseconds
  - Low jitter of less than 10 milliseconds

  If your internet connection performs badly in one or all of these areas, you will experience Quality of Service issues.

- **Ensure your LAN has sufficient bandwidth for all of your voice and data requirements**

  We don't recommend using Wi-Fi for Skype Connect. We recommend that your customer premises equipment is of standard equivalent to at least 100Mb/s CAT V.

- **Give voice data priority on your internet connection**

  Use a specialized SIP Quality of Service-based router in your corporate network which enables you to give voice data priority when leaving your corporate network and entering the internet. When internet demands are high for data, such as file downloads or email, this will prevent your voice calls being affected.

  If you are going to use the same internet connection for both your voice and office data, you will potentially have real time (voice and video) and non-time sensitive data (web browsing and email) competing for the same bandwidth. If the total available bandwidth for all data is reached or exceeded, the quality of your voice calls will be affected.

- **Give voice data priority on your LAN**

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

  Tell your corporate network to give priority to voice packets so they don't get delayed by less time-critical data packets.

4.2 Avoiding voice problems

Voice issues are typically caused by voice packets being blocked on the associated inbound or outbound port on your router or NAT device or by your ISP.

If you are using a NAT, router or firewall device in between your LAN and internet connection, ensure it does not restrict the use of the following UDP ports:

• 5060

• 8000-80xx (where x is the number of channels you require multiplied by 2)

For more help, please see your device's support documentation or contact the device manufacturer.

4.3 Simultaneous channel capacity

To use Skype Connect effectively in your business, ensure you understand your peak channel requirements for the call traffic for your business, and that you have sufficient bandwidth from your internet connection to sustain the number of calling channels required.

In simple terms, the greater number of simultaneous calling channels you wish to run, the more bandwidth you will require. Here's a table showing the approximate number of concurrent channels that can be supported using G.729 based on common internet connections:

Broadband Type	Download Speed	Upload Speed	Approximate Number of Concurrent Channels
ADSL 0.5mb	512kbs	256kbs	6
ADSL 1mb	1024kbs	512kbs	12
ADSL 8mb	8192kbs	512kbs	12
SDSL/Leased Line 2mb	2048kbs	2048kbs	48

To ensure you have sufficient bandwidth for the number of concurrent calls you wish to run and other data, it may be appropriate to consider a dedicated internet connection for your voice data.

4.4 Impact on existing data traffic and infrastructure

If you use a shared internet connection, adding Skype may impact the performance of data transfer rates for other forms of data, such as email and web browsing.

We recommend:

• **Using a Quality of Service-based router**

This will enable you to give voice priority on your corporate network so that, when your network demands are high for data, such as file downloads or email, your voice calls will be unaffected.

- **A dedicated internet connection for your voice data**

  This will ensure you have sufficient bandwidth for the number of concurrent calls you wish to run and other data.

4.5 Preparing your network to use TLS and sRTP

Transport Layer Security (TLS) and Secure RTP (sRTP) are additional Skype Connect features that encrypt SIP messages and RTP media transported between your SIP-enabled PBX and Skype, providing security from "man in the middle" attacks, such as eavesdropping, wiretapping, or threats to the security and privacy of your Skype Connect product.

TLS and sRTP are automatically enabled for Skype Connect customers. Certified equipment that has been configured according to the vendor's instructions will automatically make use of this feature. If you are unsure how to enable TLS/sRTP for your certified equipment, please refer to your vendor's support documentation or contact the manufacturer for more information.

We recommend performing the following tasks to fully prepare your network to use TLS and sRTP:

Item	Recommendation
Firewall	Configure your firewall to allow: • Outbound DNS SRV requests by Skype to the domain sip.skype.com • Receipt of DNS SRV records by Skype • Outbound TCP/IP connections from your public interface to the IP addresses provided in the DNS SRV record from Skype • Receipt of incoming TCP/IP from Skype Connect

Item	Recommendation
PBX or SIP UA (User Agent)	TLS and sRTP are automatically enabled for Skype Connect customers. Certified equipment that has been configured according to the vendor's instructions will automatically make use of this feature.
	If you are unsure how to enable TLS/sRTP for your certified equipment, please refer to your vendor's support documentation or contact the manufacturer for more information.
	Some SIP UAs require the CA-signed certificate pre-installed. If you are unsure whether this is the case, please contact Skype Customer Service for more help. For more information on how to do this, please see **7.0 Skype Connect support**.
SIP Profile	Create your SIP Profile in Skype Connect and configure your PBX using the authentication details displayed in the Registration tab (username/password).
CA certification	Upload Skype Certificate Authority (CA) certification to your mobile devices.
Device configuration	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.

For more detailed information about TLS/sRTP and Skype Connect, please see the **TLS and sRTP for Skype Connect Technical Datasheet**.

4.6 Geographic redundancy

You can set up your SIP profile to connect to Skype using either IP authentication or the DIGEST method. Both options provide primary and secondary Skype Connect authentication servers that the SIP-enabled PBX can be used for geographic redundancy. Skype' Connects authentication servers are located in different geographic locations, where in the event the primary server is not available or cannot be contacted, the SIP-enabled PBX can re-route to the secondary server using industry standard times based on DNS SRV records.

Important: When a SIP-enabled PBX fails over to the Skype Connect secondary server, calls with the primary Skype server may be dropped.

Skype Connect is not a replacement for traditional telephony services as it uses your internet connection to provide a complementary communications solution. Your SIP-enabled PBX uses your internet connection for registering to and using Skype Connect. If your internet connection drops and disconnects your SIP-enabled PBX from the internet, you will not be able to use Skype Connect.

Skype Connect is meant to complement existing traditional telephone services used with a corporate SIP-enabled PBX, not as a stand-alone solution. Skype Connect users need to ensure all calls to emergency services are terminated through traditional telephone services. Your SIP-enabled PBX should be configured to use your local fixed line telephone services, connected to the local exchange or through other emergency calling capable telephone services for access to emergency and auxiliary services.

4.7 Local Area Network (LAN) and Internet Connectivity

We don't recommend the use of Skype Connect if you have older 2Mb/s coaxial cabling or 10Mb/sec CAT V in your networks, or over Wi-Fi or other wireless networking protocols. We recommend that your communications infrastructure is of a standard equivalent to CATv(e) 100Mb/s and above.

Even if you have a dedicated internet connection for your voice traffic, we recommend that you still implement Quality of Service measures on your local LAN by using either VPNs or Quality of Service switches.

5.0 Key steps summary

The recommendations in 4.0 Technical considerations can be summarized in six key steps. Following them should dramatically improve your experience of using Skype Connect:

1. **Skype recommend you use a Skype Connect certified SIP-enabled PBX or gateway**

   Due to the complexity of SIP equipment, Skype has a certification programme to designate equipment as interoperable with Skype Connect. Skype cannot ensure all your SIP-enabled PBX features and functions will operate correctly when used with Skype Connect if the PBX or gateway used is not Skype Connect certified.

2. **Ensure you have sufficient internet bandwidth for voice and data**

   Whether you're using a dedicated or shared internet connection, ensure you have sufficient bandwidth to handle your peak concurrent calling channel requirements.

3. **Use an internet connection that has performance suitable for Skype Connect**

   Quality of Service issues are often caused by internet congestion and poor internet connection performance, causing issues such as latency, jitter and packet loss. These issues are not within Skype's control.

4. **Prioritize voice data using a Quality of Service router**

   If you're using one internet connection for both your data and voice traffic, we recommend you buy a specialised SIP Quality of Service router so you can prioritize voice traffic over time-insensitive data such as email.

5. **Consider using a dedicated internet connection for voice traffic**

   This will ensure you have sufficient bandwidth for both the number of concurrent calls you wish to run and other data.

6. **Prepare your network to use TLS/sRTP**

   This will help to ensure that your privacy is protected and that your communication services from Skype are secure provides increased protection for your communications to Skype Connect.

7. **Ensure traditional telephone services are in place for all emergency calls**

   Skype Connect is meant to complement existing traditional telephone services used with your SIP-enabled PBX, not as a stand-alone solution. You should ensure all calls to emergency services are terminated through traditional telephone services, connected to the local exchange or through other emergency calling capable telephone services.

8. **Don't run Skype Connect calls over Wi-Fi on your corporate LAN**

   We don't recommend the use of Wi-Fi connectivity for Skype Connect as call quality cannot be guaranteed.

6.0 Post-implementation tips

Once you're up and running, everything should work fine. Keep in mind the following:

- Understand that changes you make to your communications infrastructure could impact on your use of Skype Connect. If you need to make a change, plan and test it out first.

- If you experience Quality of Service issues, you'll notice very quickly.

- Monitor your call usage and set your Auto-recharge settings high enough for your employees' needs.

- If you change a piece of hardware in-between your PBX and Skype Connect, test it first.

7.0 Skype Connect support

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**.

8.0 Checklist

- ✅ I have a Skype Connect certified SIP-enabled PBX or gateway or will have one before implementing.

- ✅ I have equipment that is of 100Mb/s Cat V standard connected to my SIP-enabled PBX, and from my SIP-enabled PBX to the internet.

- ✅ If sharing internet between voice and data, I have installed a Quality of Service device between my LAN and the internet connection.

- ✅ If using a dedicated internet connection for voice, I have Quality of Service on my own internal LAN to prioritize voice packets.

- ✅ I have a Skype account.

- ✅ I have a Skype Manager account.

- ✅ I have one of Skype's accepted payment methods.

- ✅ I have visited **support.skype.com** and seen the supporting documentation for Skype Connect.

- ✅ I have regular fixed line telephone service for redundancy and emergency services access.