Primus Hosted PBX

Small Office Basic Troubleshooting Guide

Document Version 4.1

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1 Overview

This guide contains basic troubleshooting tips related to issues that Hosted PBX Small Office customers may experience.

For additional information on the Primus Hosted PBX service, please refer to the additional guides and instructional videos available on the Primus Hosted PBX support page at https://primus.ca/hpbxguide.

If you are unable to fix the problem after following the steps in this guide, your organization's registered account administrator may contact Hosted PBX Technical Support via one of the following options. Please have your account information available to you and provide it in any email support request or when you speak with a support agent.

- Call: 1-888-222-8577
- Email: businesssupport@primustel.ca
- Open a Ticket Online: https://ecare.primustel.ca

2 Things to Check and Be Aware of Before You Begin

Please ensure you are aware of the following and check available resources before you begin troubleshooting your issue.

1. Review the initial setup guides at https://primus.ca/hpbxguide and make sure your devices are getting power (power light is on), the power adapter is plugged in, and that you have all your cables plugged into the right ports.
2. Check the Primus Service Status Dashboard at http://ssd.primus.ca to see if there is scheduled or emergency maintenance that might affect your service.
3. If you recently experienced bad weather in your area and lost your power or Internet service as a result, you may have trouble getting a dial tone. Make sure your electricity and Internet have been restored, and that your Internet modem is online.
4. Some types of Internet connections are not recommended for providing reliable support for Voice over IP (VoIP) services like Hosted PBX Small Office on an ongoing basis. While the HPBX Small Office service may function with varying levels of reliability with some of the following internet connectivity methods, they are considered unsupported configurations and Primus will be unable to provide troubleshooting support if you use any of the following to connect to the Hosted PBX Small Office service:
   - Wi-Fi / Wireless
     - A Wi-Fi or Wireless LAN connection to your VoIP phone or ATA can cause call quality issues due to packet loss and jitter as a result of weak signal or interference.
   - Satellite Internet
     - Transmission of satellite Internet signals have inherent delays that cause call quality and reliability issues.
   - 3G, LTE, 4G, or Cellular Internet
     - Speed may not be an issue with some of the newer mobile Internet technologies, but latency, jitter, and packet loss can result in dropped calls and call quality issues.
     - You may also run the risk of going over your monthly mobile data usage.
   - WiMAX and Microwave
     - Similar to Cellular Internet, latency and jitter may severely impact call quality.
   - Dial-Up Internet
     - Dial-up Internet does not provide enough bandwidth to reliably support Voice Over IP (VoIP) services like Hosted PBX Small Office.

Remember, the quality of your Internet connection makes a big difference!

The call quality of your VoIP phones is only as good as the quality of your Internet connection. Therefore, having a stable, high-quality Internet connection with sufficient bandwidth is essential.

The real-time nature of phone calls also means that you may not notice any Internet connectivity issues with your computers, but you may still have issues with voice calls. Voice calls are more sensitive to network problems which can result in call quality and connection issues even when other aspects of your network seem fine.

3 Issue Identification

Identify your issue below and follow each of the troubleshooting procedures in the order listed. The issues are grouped into the following sections:

- Issues Related to Specific Devices
- Dialing or Call Connection Issues
- Audio Quality Issues

3.1 Issues Related to Specific Devices

3.1.1 Linksys/Cisco Analog Telephone Adapter (ATA)

Refer to the troubleshooting procedures for the Linksys/Cisco Analog Telephone Adapter (ATA).

3.1.2 Panasonic TGP550 and TPA50

Refer to the troubleshooting procedures for the Panasonic TGP550 and TPA50 phones.

3.1.3 Panasonic TGP600, TPA60, and TPA65

Refer to the troubleshooting procedures for the Panasonic TGP600, TPA60, and TPA65 phones.

3.2 Dialing or Call Connection Issues

3.2.1 No Dial Tone or Fast Busy Signal

If you experience no dial tone, or a fast busy signal after you pick up the phone, check the following items in order:

1. Check Cabling
2. Check Your Internet Connection
3. Power Cycle your Devices
4. Plug your VoIP Device Directly to your Internet Modem
5. Review Router/Firewall Settings

3.2.2 No Ringing After Placing Call or No Audio or One-Way Audio

If you have dial tone, but experience any of the following:

1. No ringing after dialing the number
2. No audio after the call is made
3. One-way audio after the call is made
   - One-way audio is when one party on the call can hear the other, but cannot be heard by the other party

To troubleshoot any of the above conditons, check the following items in order:

1. Plug your VoIP Device Directly to your Internet Modem
2. Review Router/Firewall Settings

3.2.3 Unable to Receive Calls

If you are unable to receive calls that you know were sent to your phone number, check the following items in order:

1. Plug your VoIP Device Directly to your Internet Modem
2. Review Router/Firewall Settings
3. Call your Phone Number from a Different Device or from a Land Line

3.2.4 Unable to Call a Specific Number

If you are unable to call a specific number, check the following items in order:

1. If it is a long distance or toll-free number, make sure you dial "1" before dialing the number
2. Make sure it is not an issue with the destination number. Try calling the number using a different device or cell phone. If you have verified that there is no issue with the destination number, note down the time and date of your attempted call, which device you used, the destination number, and have your registered account administrator contact Primus Hosted PBX Technical Support.

3.3 Audio Quality Issues

3.3.1 Dropped Calls

If you are experiencing calls that terminate (or drop) unexpectedly in the middle of a conversation, check the following items in order:

1. Check Cabling
2. Check Your Internet Connection
3. Run a Network Test
4. Power Cycle your Devices
5. Plug your VoIP Device Directly to your Internet Modem
6. Review Router/Firewall Settings

3.3.2 Voice Delays

If you are experiencing long delays between people talking on a call, check the following items in order:

1. Run a Network Test
2. Plug your VoIP Device Directly to your Internet Modem

3.3.3 Robotic, garbled, or distorted sound

If you are experiencing robotic, garbled or distorted sound when talking on a call, check the following items in order:

1. Check Cabling
2. Try a Different Headset, Handset, or Speaker
3. Run a Network Test
4. Plug your VoIP Device Directly to your Internet Modem
5. Avoid Internet Load Sharing

3.3.4 Gaps in speech, choppy speech, or voices cut in and out

If you are experiencing gaps in speech, choppy speech or people's voices cut in and out when talking on a call, check the following items in order:

1. Run a Network Test
2. Power Cycle your Devices
3. Plug your VoIP Device Directly to your Internet Modem
4. Review Router/Firewall Settings

3.3.5 Echo Throughout The Call

Echo is typically caused by audio feedback from the speaker to the microphone. To reduce audio feedback, try turning down your speaker volume to see if there is a difference. It may also happen on low-quality or aging phone sets and headsets, or headsets that are not optimally positioned. Occasionally, it may also be related to electrical feedback from faulty phone wiring.

The problem may also be caused by the person on the other end of the connection. Try making a call to another device or person and see if the problem still exists.

3.3.6 Constant Humming or Static

A low frequency hum may be caused by interference from power transformers, power lines, power supplies, or other electrical equipment that are close to a cable or the IP device that the voice path is running on. It may even be caused by radios, microwaves, satellite dishes, wireless routers, cell phones, televisions, or surge protectors.

Move any electronic devices that could be causing interference from your IP device or any cables connected to it.

You can also try a different speaker or headsets. Poor quality and aging headsets are known to degrade sound quality or introduce additional background noise.

3.3.7 Buzzy Voice

A buzzy voice quality issue is usually caused by speakers that are turned up too high, causing the sound to buzz at higher volumes. Turning the volume down usually eliminates the problem.

3.3.8 Hollow or tunnel-like sound quality

This is when the audio sounds like you're in a tunnel or sounds a bit hollow. This is actually caused by a low delay echo and may be associated with the internal feedback of the particular handset or headset you are using. Consider using a different device.

4 Troubleshooting Procedures

The sections below will outline the steps for each troubleshooting procedure. Please refer to the previous section to first identify your issue. Once you identify your issue, follow the list of steps in the appropriate troubleshooting procedure.

4.1 Linksys/Cisco Analog Telephone Adapter (ATA)

If you are having issues with your Linksys or Cisco ATA device, check the following items to help troubleshoot the issue.

- The ATA has a built-in router. Therefore, we do not recommend connecting the ATA between your Internet modem and router, as it may introduce a Double NAT setup on your network.
- If you have both a VoIP phone and an ATA, Primus does not recommend connecting the ATA to the additional Ethernet port at the back of the VoIP phone, or vice versa. This configuration may cause unexpected issues.
- If your analog phone is connected to the ATA via a phone jack in your wall, try plugging the phone directly to the ATA instead. If this resolves your issue, then your problem may be related to your internal wiring or phone jack.
- Make sure your telephone is set to its tone setting, not pulse.

4.1.1 ATA Status Light Indicators

Light Icon	Light Status			
	Solid Green	Flashing Green	Flashing Red	No Light
Power	ATA is powered on and connected to the Internet	The ATA is attempting to connect to the Internet. If this does not turn solid, verify your Internet connection is functioning properly.	ATA is attempting a self-test. If it is flashing indefinitely, the ATA may be defective.	ATA is off and not powered.
Internet	Internet connection has been established. There is no Internet traffic.	Internet connection has been established and data is being sent and received.		No Internet connection has been established.
Phone 1 Phone 2	Telephone or fax machine is active and connected to the HPBX Small Office system.	Phone or fax is in use or off-hook.		Port not ready and is not connected to the HPBX Small Office system.

4.1.2 Calls From ATA Connect - Computer Connected To ATA Can't Get To Internet

If calls from the ATA connect fine, but the computer connected to the ATA cannot connect to the Internet, please check the following items in order:

1. Ensure the computer is connected to the port labelled "Ethernet" on the back of the ATA.
2. Check Cabling
3. Power Cycle your Devices
4. Check your computer configuration – can your computer connect to the Internet if it is plugged directly to your modem or router?

4.1.3 Using a Fax Machine

HPBX Small Office supports the T.38 fax codec for Fax over IP networks. T.38 requires about the same bandwidth as an HPBX Small Office voice call. However, fax is more sensitive to network issues such as delay, jitter, and packet loss. Where a voice call may have a moment or two of robotic or poor voice quality, the same event may be enough to disconnect a Fax over IP transmission. Therefore, having a high quality and stable Internet connection is critical for Fax over IP to work. Due to the potential for momentary periods of jitter, delay and packet loss on most Internet access connections, Primus only supports Fax as a best effort service and cannot guarantee successful transmission every time.

You should not use a fax machine with an ATA if you do a high volume of faxes or multi-page faxes on a daily basis, or if faxing is critical for your business (ex.: you send and receive medical, legal, or sales documents on a daily basis). If you primarily receive faxes and it is not critical for your business, Primus recommends the inbound fax-to-email service available on HPBX. If you send or receive a high volume of faxes, Primus recommends using a regular analog phone line connected to your fax machine instead. Please contact your Primus Account Manager for details.

Using T.38, HPBX Small Office supports the V.29 and V.17 fax standards at a baud rate or data rate of up to 9600 bits per second. Although V.17 also supports a 14,400 bits per second data rate, it is not recommended, as it increases the failure rate of fax transmissions. Please make sure your fax machine is configured to one of the compatible settings.

If you still have issues after configuring your fax machine to the right settings, you may also want to try turning Error Correction Mode (ECM) on your fax machine on or off. With ECM on, the image quality of your fax will be better, but it may increase both the time it takes to send the fax, and the chances of faxes sent unsuccessfully. Turning ECM off may improve success rates, but any missing data will result in parts of the fax image being omitted.

Here are some additional steps you can try to improve Fax connectivity:

- Do not share the line with additional phones. Always connect the fax machine directly to the ATA and do not split the line.
- Try lowering the baud rate to 7200 or less.

4.2 Panasonic TGP550 and TPA50 Phone

If you are having issues with your Panasonic TGP550 or TPA50 devices, check the following items to help troubleshoot the issue.

- If you can make calls from the Panasonic corded base unit, but not from a cordless handset, please ensure the handset is paired with the base unit and that you have assigned it to an HPBX Small Office user. The *"Panasonic KX-TPA50 Additional Handset Quick Set-Up Guide"* and *"Panasonic TGP550/TPA50 Handset User Assignment Guide"* are available on the Primus Hosted PBX support page at https://primus.ca/hpbxguide and will outline the steps needed to perform these tasks.
- If you have both the TGP550 Panasonic phone and an ATA, Primus does not recommend connecting the ATA to the additional Ethernet port at the back of the Panasonic base unit, or vice versa. This configuration may cause unexpected issues. Please connect both units directly to your Internet modem or router.
- If you have reception or voice quality issues when using the cordless handset, try charging the phone's battery, or move closer to the corded phone base station for better reception. Note, the charging station for the cordless handset is only for charging the battery and is not used for wireless reception.
- Please review the Panasonic TGP550 and TPA50 Manufacturer Guides for additional information and troubleshooting tips available on the Primus Hosted PBX support page at https://primus.ca/hpbxguide.

4.2.1 Status Light Indicators

Status Indicator	Description
Solid Green	The base unit is connected to the internet and connected to the HPBX Small Office system.
Flashing Green	The base unit is downloading data. Do not disconnect the Ethernet cable or AC power adaptor from the base unit until the light becomes solid green.
Solid Red	The base unit is booting up. This will take about 40 seconds.
Flashing Red	The base unit is registering a handset.
Solid Yellow	The base unit's IP address may conflict with another IP address on your network.
Slow Flashing Yellow (once per second)	The base unit is attempting to connect to the Internet or to the HPBX Small Office system. Please wait. If it flashes indefinitely, check your network settings, ensure your Internet connection is working, or try restarting all your devices.
Quick Flashing Yellow (4 times per second)	Unplug the base unit's AC adaptor to reset the unit, then reconnect the AC adaptor. If it is still flashing, there may be a problem with the base unit hardware. Contact Primus HPBX Technical Support.
Off	The base unit is off and not powered, the Ethernet cable is not connected properly, or it is not getting an active network connection. Ensure your Internet connection and network devices are turned on.

4.3 Panasonic TGP600, TPA60, and TPA65 Phone

If you are having issues with your Panasonic TGP600, TPA60 or TPA65 devices, check the following items to help troubleshoot the issue.

- Ensure the unit is powered on. For the TPA60 handset, you can turn the handset on and off by pressing and holding the power/cancel button.
- If you can make calls from one of the Panasonic cordless handsets but not from any of the other cordless handsets or wireless desk phones, please ensure the other handsets or phones are paired with the base unit, and that you have assigned it to a HPBX Small Office user. The *"Panasonic KX-TGP600, KX-TPA60, and KX-TPA65 Quick Set-Up Guide"* and *"Panasonic TGP600, TPA60, and TPA65 Phone User Assignment Guide"* are available on the Primus Hosted PBX support page at https://primus.ca/hpbxguide and will outline the steps needed to perform these tasks.
- If you are having reception or voice quality issues:
  - Try charging the phone's battery for the cordless handset
  - Hold the bottom half of the cordless handset when in use. The antenna is located in the upper half of the handset.
  - Obstacles such as walls, high metal shelves, and reinforced concrete walls will restrict your operating range. Try avoiding these obstacles or move closer to the base unit for better reception. Check the signal strength indicator on the display of your cordless handset or wireless desk phone. Note, for the cordless handset, the charging station is for charging the battery only and is not used for wireless reception.
  - Occasional noise or interference may occur due to electromagnetic radiation from objects such as refrigerators, microwave ovens, faxes, TVs, radios, personal computers, fluorescent lamps, motors, heating appliances, and other electrical equipment. If noise is noticeable, keep the cordless handset or wireless desk phone away from these devices.
  - Repeaters are available to extend the coverage area of your base unit. Contact your Primus Account Manager for details.
- Please review the Panasonic TGP600, TPA60, and TPA65 Manufacturer Guides for additional information and troubleshooting tips available on the Primus Hosted PBX support page at https://primus.ca/hpbxguide.

4.3.1 Status Light Indicators

Status Indicator	Description
Solid Green	The base unit is connected to the internet and connected to the HPBX Small Office system.
Slow Flashing Green (once per second)	The base unit is being used for a call. or The base unit is downloading data. Do not disconnect the Ethernet cable or AC power adaptor from the base unit until the light becomes solid green.
Quick Flashing Green (4 times per second)	The handset/desk phone is busy.
Solid Red	The base unit is booting up. This will take about 40 seconds.
Slow Flashing Red (once per second)	The base unit is registering a handset/desk phone.
Quick Flashing Red (4 times per second)	The base unit is paging handsets/desk phones (by using the handset locator button).
Solid Yellow	The base unit's IP address may conflict with another IP address on your network.
Slow Flashing Yellow (once per second)	The base unit is attempting to connect to the Internet or to the HPBX Small Office system. Please wait. If it flashes indefinitely, check your network settings, ensure your Internet connection is working, or try restarting all your devices.
Quick Flashing Yellow (4 times per second)	Unplug the base unit's AC adaptor to reset the unit, then reconnect the AC adaptor. If it is still flashing, there may be a problem with the base unit hardware.
Slow switching (Red → Green → Amber → Off)	The base unit is in maintenance mode. Once it is off, turn the base unit back on again.
Quick switching (Red → Green → Amber → Off)	The base unit is restarting. Please wait.
Off	The base unit power is off. or The Ethernet cable is not connected properly. or Your network devices (modem, switch, router, etc) are turned off or not functioning.

4.4 Check Cabling

A bad or loose Ethernet or phone cable can sometimes cause speech to be broken up or distorted.

- Make sure cables are all snugly plugged into the right ports. Unplug and reconnect each end of the cables, ensuring the connectors click and lock into place when inserted.
- Try switching to a different cable to see if it makes a difference.
- Ensure your cables are not excessively long. For example, CAT5 cables should be no longer than 100 metres.

4.5 Check Your Internet Connection

Make sure your Internet connection and associated modem or router is working and does not have intermittent problems. To make sure other devices on your network aren't causing issues to your Internet connectivity, you may want to unplug or turn off all devices on your network except for a single computer, and see if that computer can consistently access the Internet.

Also, make sure your modem or router is functioning and automatically assigning an IP address to devices plugged into your network.

4.6 Run a Network Test

Go to http://speedtest.primus.ca and run the test by following the instructions on the site. You may wish to run multiple tests over a period of time to compare results as the test will only capture how your Internet service is performing while the test is conducted.

Each simultaneous VoIP call will consume approximately 100kbps in both the upload and download direction. You will need to ensure that your internet service is fast enough in both the upload and download direction to support the number of simultaneous VoIP calls you need to place, along with any other internet traffic being generated.

In general you want to ensure that latency is less than 150ms and jitter is less than 30ms. Latency and jitter values higher than these will certainly cause call quality issues, and the lower the number, the better.

If the latency and jitter parameters are above those values, or your upload and/or download speeds are not sufficient to handle a VoIP calls, it means either your connection or your network usage is inadequate to support VoIP reliably. You can improve voice quality by reducing the amount of traffic on your network. Reduce streaming of audio/video files or TV, minimize large file downloads, or reduce the number of people or devices using your Internet connection.

In some cases, your Internet connection may inherently be introducing packet delays and jitter due to the way your Internet Service is routing traffic. You may need to switch to a higher quality Internet connection or change to an Internet provider that can better support VoIP.

For more information, call Primus Hosted PBX Technical Support and have your test results available.

> NOTE: Primus can only provide limited support regarding your local network and your router and firewall, since it is not managed or maintained by Primus. If you do not have access to your router or firewall, or are unfamiliar with your network, contact your IT support for assistance.

4.7 Power Cycle your Devices

Power cycle your modem, router, firewall, VoIP phones, or analog phone adapters in sequence by following the steps below:

1. Turn off or disconnect the power from your Internet modem, firewall/router/switch, analog phone adapters, and VoIP phones.
2. Leave the devices off or disconnected from power for at least one minute.
3. Turn on your Internet modem. Wait until the modem has completed its power-up sequence (all the indicator lights are on as normal).
4. Turn on your firewall/router/switch and wait until the device has completed its start-up process.
5. Turn on your analog phone adapters or IP phones and wait until it has fully booted up.

If you have a computer connected to the Internet, open your web browser and check your Internet connection by visiting any website, such as https://primus.ca.

4.8 Plug your VoIP Device Directly to your Internet Modem

If you have your VoIP phone or analog phone adapter plugged into a switch, router, or firewall, try plugging the device directly to your Internet modem instead. Make sure no other devices are on the network. If the device works, one of your network devices may be causing the issue due to a configuration issue or malfunction.

4.9 Avoid Internet Load Sharing

Load sharing is when your Internet connectivity consists of multiple Internet connections aggregated using a dual-WAN router. This setup can cause a lot of jitter and audio quality problems, especially if voice packets can go through either connection, and may cause packets to arrive out of order. It is best to configure load sharing so that VoIP calls go through one consistent route at a time, instead of spreading packets from the same call over different paths, or avoid using load sharing altogether.

NOTE: Primus can only provide limited support regarding your local network and your router and firewall, since it is not managed or maintained by Primus. If you do not have access to your router or firewall, or are unfamiliar with your network, contact your IT support for assistance.

4.10 Call your Phone Number from a Different Device or from a Land Line

Do you find that callers from a particular carrier (mobile or landline) are not able to call your HPBX Small Office phone number?

Some providers are slow to update their database with new phone numbers so that they can route calls correctly for new exchanges.

If this is the case, Primus can contact the carrier in question to have them update their database (called LERG information). You will need to provide Primus with the following information:

- The phone number and carrier of the caller
- Date and time of the last call
- City or location of the caller when making the call (for mobile carriers, it helps them locate the specific switch that needs to be updated)
- The message or recording received when making the call (busy tone, or “Number is not in service,” etc.)
- The HPBX Small Office phone number the caller is trying to reach

4.11 Try a Different Headset, Handset, or Speaker

You may have a faulty speaker, headset, or handset. Try using a different device on that same connection and see if it makes a difference. Poor-quality headsets are also known to degrade sound quality. If using cordless phone, make sure battery is fully charged. You may want to try charging the wireless phone's battery, changing channels, or move closer to your wireless phone base station.

4.12 Review Router/Firewall Settings

The recommended setup for a good voice connection is to always connect your Internet modem directly to the VoIP device provided by Primus. However, if your network requires using a router or firewall, then you will need to make sure your device is properly configured to allow VoIP to function properly.

NOTE: Primus can only provide limited support regarding your local network and your router and firewall, since it is not managed or maintained by Primus. However, here are some steps you can take to troubleshoot your issue. Note, these steps require a basic knowledge of computer networking. If you do not have access to your router or firewall or are unfamiliar with your network, we recommend contacting your device vendor or IT support for assistance.

4.12.1 Does your network have two routers?

One common cause of router issues affecting VoIP is a Double NAT setup. Network Address Translation (NAT) is used by the router to direct traffic to your computer on your local network. A Double NAT situation occurs when two routers are placed one after another, which can happen if a router is added to the network that already has a modem/router combination device.

A computer behind a Double NAT setup can still browse the web, but certain applications, such as email and VoIP may not work at all. To resolve this, either remove the additional router from your network, or turn off the router functionality on that device. (Some routers allow you to disable the routing functionality so that it just functions as a switch.) Alternatively, if you have a modem/router combination device, you can configure the device in "bridged" mode to turn off the router functionality on your modem so that the NAT is only performed by your standalone router.

If you use DSL Internet and have your modem configured in "bridged" mode, you will need to input your DSL PPPoE credentials to your standalone router so that you can connect to the Internet. Some DSL modems/routers support a "half-bridged" mode, where the modem will perform the PPPoE login authentication for DSL connectivity, but leave the NAT to your standalone router. Contact your IT support or your Internet Service Provider for assistance.

NOTE: If you have an Analog Telephone Adapter (ATA) from Primus, the ATA also has a built-in router. Therefore, avoid connecting a router to the ATA's Ethernet port, as that will cause a Double NAT situation for any devices subsequently connected to the router.

4.12.2 Make sure your router has assigned an IP address to your VoIP phone or ATA

Most routers are configured by default to use DHCP to automatically assign a private IP address to a device connected to the network. To check if an IP address has been assigned to your VoIP device, log into your router to see what devices are on the network. Most routers allow you to log in to their control panel through a web browser and going to 192.168.0.1 or 192.168.0.100. Find the list of devices on the network and make sure the MAC address of your VoIP phone or ATA (usually written as something like 00:04:F2:B0:69:E5, labelled under the device) matches one listed by the router.

4.12.3 Update your firmware

Updating the firmware of your router or firewall may sometimes resolve your issues. Please refer to your device manufacturer for instructions and possible risks.

4.12.4 Setup Quality of Service (QoS) on your Router

Most routers have a setting for Quality of Service (QoS) or bandwidth management, which can help prioritize VoIP traffic over regular Internet traffic and improve call quality. Some routers dedicate a specific amount of bandwidth to VoIP when it's in use. Configuring QoS or bandwidth management is different for each router – refer to your device manufacturer for more information.

Note that configuring QoS will only prioritize outbound traffic from your network to the Internet. Incoming Internet traffic to your network will not be prioritized by your Internet Service Provider, as no provider currently supports end-to-end QoS on a regular, high-speed Internet connection. The only way to guarantee end-to-end QoS is to use a private, dedicated connection that connects your VoIP devices directly to the Hosted PBX servers – this option is available by upgrading from HPBX Small Office to a HPBX solution with a managed voice network and access connection. Contact your Primus Account Manager for more information.

4.12.5 Ports and IP Addresses used for HPBX Small Office

Do you experience any one of the following symptoms?

- One-way audio or no audio
- Registration problems (device won't connect to HPBX Small Office system)
- The VoIP connectivity status light on your device goes out intermittently or constantly
- When making outbound calls, you hear a busy signal even though the person you are calling hears a ring
- You hear a busy signal in the middle of the call
- Your calls have a maximum duration of exactly 10 or 30 minutes

If so, your router or firewall may be preventing the VoIP signalling from working properly. For example, your firewall may be allowing the VoIP signalling to pass through to establish a call, but not allow the voice (RTP) packets to pass through.

If you have a firewall (usually built into your router) and have restrictive security in place, please ensure the following ports and IP addresses are accessible on your network.

> NOTE: The following information requires a basic knowledge of computer networking. Please contact your IT support or router/firewall manufacturer for assistance, as Primus will not be able to provide support or make changes to your own firewall or other network devices.

Primus uses the following ports and they must be open on your network for HPBX Small Office to function:

Protocol	Use	Inbound	Outbound
UDP	Used for RTP voice packets. When a call is made, random ports between the range listed are used to carry the conversation. If any of these ports are blocked, you may experience one way or no audio.	Any	1024-65534
TCP	Used for SIP signalling (for setting up VoIP calls)	5060 or 5065	5060 or 5065

The above ports must be available for the following IP addresses and URLs:

- 209.183.11.200
- 209.90.170.40
- 216.254.141.228
- 209.183.11.198
- 209.183.11.199
- 209.183.11.200
- 209.90.170.40
- obp-01.bvoice.primus.ca
- vsbc-hpbx-1.voip.primus.ca

4.12.6 Other Troubleshooting Tips

- Turn off SIP Application Level Gateway (SIP ALG)
  - SIP ALG is a feature in most firewalls that is meant to fix or improve VoIP applications on the network. However, the feature tends to cause more issues than it fixes and may interfere with HPBX Small Office functionality.
  - This feature is usually turned on by default. We recommend turning it off if you are having any issues.
- Turn off Stateful Packet Inspection (SPI)/Dynamic Packet Filtering
  - SPI is a feature that ensures all inbound packets are a result of a request from your network and is designed to prevent harmful or unrequested packets from entering your network. However, this may interfere with the proper functioning of VoIP. Try turning it off and see if it makes a difference.
- Turn off UPnP (Universal plug-and-play)
  - Some older routers with UPnP enabled may interfere with VoIP. Try turning it off and see if your problem is resolved.
- Enable RTP and SIP
  - Make sure your firewall allows RTP and SIP traffic.
- Disable any other VoIP-specific settings on your firewall or router
  - Some network devices have additional settings for VoIP services. Try enabling or disabling those features and see if it makes a difference.

- Assign a public IP address to your device
  - Obtain multiple public static IPs from your Internet Service Provider (your provider may charge for this feature), and assign a public IP address to each of your Primus VoIP devices. This should eliminate any issues caused by NAT (Network Address Translation).
- Establish a Demilitarized Zone (DMZ) on your network
  - Enable the DMZ option on your router, which will open the firewall for specific devices on your network. Add your Primus VoIP devices to the DMZ. Note, if there are other devices that need firewall protection, but also need to establish connections to the Internet, this may not be a viable option.
- Port Forwarding/Port Mapping/Port Triggering
  - Port forwarding directs all data received on a specified port number to one specific device on your private network. To set up port forwarding, you need to assign a static IP address to your VoIP device. Ensure the all the ports listed in the table above are configured to forward to your VoIP device.

NOTE: If you have multiple Primus VoIP devices on your network, port forwarding will not be a viable solution.

- Contact your router/firewall manufacturer
  - Support numbers for some common router/firewall manufacturers are listed below. Please visit your manufacturer's website for more details.
    - D-Link - 1-800-354-6522
    - Linksys - 1-800-326-7114
    - Cisco - 1-800-553-2447