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WOW! Business Services

Hosted VoIP Troubleshooting

User Guide v2.0

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Purpose

Our goal is to deliver high-quality voice services to streamline your operations so you can focus on your core business. It is important to understand your network requirements in support of your new Hosted VoIP services. This User Guide provides basic guidelines to assist with integrating these services with your existing network. It also includes many Frequently Asked Questions (FAQs) as well as some basic troubleshooting recommendations for your to reference.

General Recommendations

To maintain consistent voice call quality, it is important that the servicing Internet connection is stable. If there are any reliability issues with that connection (ie excessive Jitter, Packet Loss, etc) during general usage, it will cause poor audio on WOW!'s Hosted VoIP service. Also, while the required bandwidth for a single VoIP call is only approximately 100 Kbps, voice quality issues can occur if a network is excessively utilized or improperly configured. Here are some important requirements to consider:

- Internal wiring should be well sorted and consistent, preferably using shielded cables to reduce RF interference within the building. If this is an issue, you may experience general service issues.
- The goal is to have 0% packet loss on the internal network. Depending on the equipment being used to provide Internet service to WOW!'s devices, this may or may not be an available metric.
- <20ms jitter. We recommend <20ms jitter, but this measurement may also not be available.
- As mentioned, a single VoIP call only utilizes approximately 100 Kbps, but it is important to understand overall call volumes as well as the bandwidth available to support these voice services. Keep in mind excessive data usage on your network could negatively impact voice quality.
- If WOW!'s Hosted VoIP service is being deployed within an already complex network with multiple VLANs, please ensure that the Hosted VoIP service resides within its own VLAN. This VLAN should have Internet connectivity that meets the above guidelines.

Specific Feature Interactions

All Hosted VoIP features have been successfully tested and verified by WOW! in our lab. However, it is possible that certain CPE may exhibit issues when certain features are in use. Phones using side cars with multiple monitored extensions or ring all hunt groups, for example, could potentially have service issues. To minimize this, it is recommended to follow the guidelines below when configuring the servicing router/firewall for use with WOW!'s OTT Hosted VoIP services.

It should also be noted that paging services will only function with devices local to the OTT device (i.e. the same physical network) and with that device fully supporting multicast traffic. For the purposes of paging, Poly phones will need to use the 224.0.1.116 network for multicast.

Router/Firewall Considerations

There are numerous routers and firewalls available in the market today. This section serves as a general guideline for configurations. We do not have specific examples for each manufacturer and device as some settings may vary.

If the sole device in use is your internet service provider's modem with its built-in router and switch, please consult that provider's documentation to verify its configuration.

Enabling SIP ALG

SIP ALG is a feature built into many routers and firewalls to help manage and simplify the setup of SIP-based communications.

- **Simplifies NAT Traversal:** SIP ALG helps resolve issues related to Network Address Translation (NAT), ensuring that SIP traffic can traverse routers without dropping calls or experiencing one-way audio.
- **Automatic Port Management:** It dynamically opens and manages the necessary ports for SIP signaling and media streams, reducing the need for manual port forwarding.
- **Packet Inspection and Translation:** SIP ALG can inspect SIP packets, modify headers as needed (e.g., replacing private IP addresses with public ones), and ensure proper routing between endpoints.
- **Improved VoIP Call Reliability:** By managing SIP signaling and media flow, SIP ALG can improve the reliability of VoIP calls, particularly in networks with multiple NAT layers or complex configurations.
- **Ease of Deployment:** It simplifies VoIP setup by reducing the need for advanced network configuration, making it beneficial for small businesses or environments with limited IT resources.

Please Note: While SIP ALG offers these benefits, it can also introduce issues in certain scenarios, such as incorrect header modification or compatibility problems.

Quality of Service (QoS)

- Prioritize SIP and RTP ports
- Assign high priority or dedicated bandwidth for VoIP traffic
- DSCP values - SIP (signaling) 24
- DSCP values - RTP (media) 46

Why: Ensures call quality by reducing jitter, latency, and packet loss during congestion

Firewall Rules

- Allow SIP and RTP traffic.
- Restrict external access to only the provider's "WOW" IPs for security.

Why: Minimizes unauthorized access and SIP attacks.

FQDN/IPs Per Market and Ports Used from WOW!

Our SIP services use UDP ports for both incoming and outgoing packets. The labeled ports below reference the available range from WOW! by market.

Market	FQDN	Public IP	Port Range
Huntsville	alhunt.hvoip.voice.wowway.com	75.76.34.10	16384 to 65535
Montgomery	almont.hvoip.voice.wowway.com	75.76.34.26	16384 to 65535
Panama City	flpana.hvoip.voice.wowway.com	75.76.38.10	16384 to 65535
Pinellas	flpine.hvoip.voice.wowway.com	75.76.38.26	16384 to 65535
Augusta	gaagst.hvoip.voice.wowway.com	75.76.11.42	16384 to 65535
Columbus GA	gaclmb.hvoip.voice.wowway.com	75.76.11.26	16384 to 65535
Michigan	mitroy.hvoip.voice.wowway.com	75.76.69.10	16384 to 65535
Charleston	scchar.hvoip.voice.wowway.com	75.76.36.10	16384 to 65535
Knoxville	tnknox.hvoip.voice.wowway.com	75.76.36.26	16384 to 65535

Port Availability – ISP – Firewall and ISP

Please Note: It is possible (albeit unlikely) that your Internet Service Provider (ISP) may block certain destinations or ports for security or for commercial reasons. The above destinations must be reachable for WOW! voice services to function correctly. Your firewall and your ISP must allow these destinations for your market.

WOW! SIP Provisioning System - Testing Connectivity

For phones and soft clients to be able to receive their configuration, they must be able to connect to WOW!'s SIP Provisioning system. Port 5060 will be used for IP phones, port 443 for the WOW! Mobility client and port 80 for the desktop client.

If devices are having issues, check that your DNS can reach the following:

- voice.wowway.com (via HTTPS, port 443)
- phone.wowforbusiness.com (via HTTPS, port 443)

To test DNS, open a command prompt or terminal session on a device on your network, and type (for example) 'nslookup voice.wowway.com' and verify that an IP address is returned (and not an error message).

To test connectivity, open a web browser on a device on your network and attempt to go directly to the above two domain names. The **https.voice.wowway.com** should return a 404 error message and phone. wowforbusiness.com should return a login page. If they are blocked, you will see connection errors.

You can also open a command prompt or terminal session on a device on your network and type 'telnet inevvl.hvoip.voice.wowway.com 5060' and verify that a connection occurs. If the telnet session hangs up while trying to connect, or you get rejected immediately, there is an issue. You should see "connected to" or see the window go to a blank screen with a blinking cursor if a connection is successfully made. You can then use ctrl+] or ctrl+c to end the connection.

Poly/HP ZTP – Ports and Protocols

Destination	Type	Port	Purpose	Protocol
aadcdn.msftauth.net login.live.com	TCP	443	Microsoft SSO	HTTPS
accounts.google.com lh3.googleusercontent.com	TCP	443	Google SSO	HTTPS
cdn.polycom.com	TCP	443	Device Software	HTTPS
global.azure-devices-provisioning.net ztp-iot-prod-dps-eastus2.azure-devices-provisioning.net ztp-iot-prod-dps2-eastus2.azure-devices-provisioning.net ztp-iot-prod-dps3-eastus2.azure-devices-provisioning.net ztp-iot-prod-dps4-eastus2.azure-devices-provisioning.net	TCP	443	Policy & Cloud Settings, Device Status, Usage Telemetry	AMQP-WS
login3.id.hp.com	TCP	443	HP SSO	HTTPS
prov.obitalk.com	TCP	443	Zero Touch Provisioning (Obihai Devices)	HTTPS
swupdate.lens.poly.com	TCP	443	Device Software	HTTPS
ztp-iot-prod-iothub-eastus2.azure-devices.net ztp-iot-prod-iothub2-eastus2.azure-devices.net ztp-iot-prod-iothub3-eastus2.azure-devices.net ztp-iot-prod-iothub4-eastus2.azure-devices.net	TCP	443	Device Active Management	AMQP-WS
ztp.polycom.com	TCP	443	Zero Touch Provisioning	HTTPS

Adjusting NAT Timeout & Fast NAT Settings

1. **Maintaining SIP Session Stability:** SIP relies on maintaining session states for call signaling and media streams. Increasing the NAT timeout ensures that the NAT entry stays active longer, preventing call drops due to expired sessions.
2. **Minimized Reconnection Delays**: Prolonging NAT timeout reduces the need for frequent re-registration or reconnection, ensuring smoother VoIP operations
3. **Improved Packet Processing Speed:** Fast Nat allows for quicker handling of SIP traffic by reducing the overhead of deeper packet inspection, resulting in lower latency for VoIP calls.
4. **Reduced CPU Load on the Router:** Fast Nat optimizes routing efficiency, which is especially useful in networks handling a high volume of SIP calls
5. **Timeouts too short:** Can lead to dropped calls or disrupted sessions
6. **Timeouts too long:** May hold unnecessary resources and potentially expose security vulnerabilities.
7. **Fast Nat limitations:** Might bypass some SIP-specific functions, like ALG, that depend on deeper packet processing

NAT Settings (pinhole timeout)

This refers to the amount of time a network device like a router or firewall will keep a specific port open for incoming traffic after the last communication on that port.

- **UDP Timeout:** 60-120 seconds or higher
- **TCP Timeout:** 300 seconds or higher (*NOTE - We do not currently offer SIP over TCP for Hosted VoIP services.*)

Why should we change these values? To prevent premature session closure and ensure NAT mappings stay active during calls.

WOW! will tear down any call if all media streams carry no media (no RTP or RTCP) for 30 seconds, unless the streams are placed on hold.

Troubleshooting

Reported Problem	Possible Issue
IP Phone will not power on	Not receiving power from either POE Switch or POE injector
IP Phone cannot be configured from factory default	Does the device currently have IP from the local LAN
	Does the local LAN have access to the Public Internet
	Does the device have access to DNS access
IP Phones randomly rebooting	These devices resync their configuration between 2am-6am (local time) If a new config is sent during this window the device could reboot. Confirm the date & time are correct on the device
	If there are loose connections (Ethernet/external power supply) this could cause the issue
IP Phone is unable to receive provisioning updates	Does the local LAN have access to the Public Internet
	Does the device have access to DNS access
Long active calls drop	This could be related to a NON SIP aware router that is incorrectly marking the voice packets, causing NAT sessions to close too fast.
Enhanced Features (Call Park, Monitored Extension) are not functioning correctly	This could be related to a NON SIP aware router ip incorrectly marking the voice packets & due to that this the NAT pinhole via the router are closing to fast
Degraded / Poor Call Quality	Insufficient bandwidth: Not having enough upload and download speed on your internet connection to handle VoIP traffic
	High latency: Significant delay in data transmission, causing noticeable pauses in conversation
	Network jitter: Irregular packet arrival times leading to choppy audio.
	Packet loss: Data packets failing to reach their destination, resulting in gaps in audio
	Quality of Service (QoS) settings on your router not set to prioritize VoIP traffic correctly.
	SIP ALG settings may need to be adjusted or disabled.
	Physical connection issues such as loose cables or damaged network infrastructure.

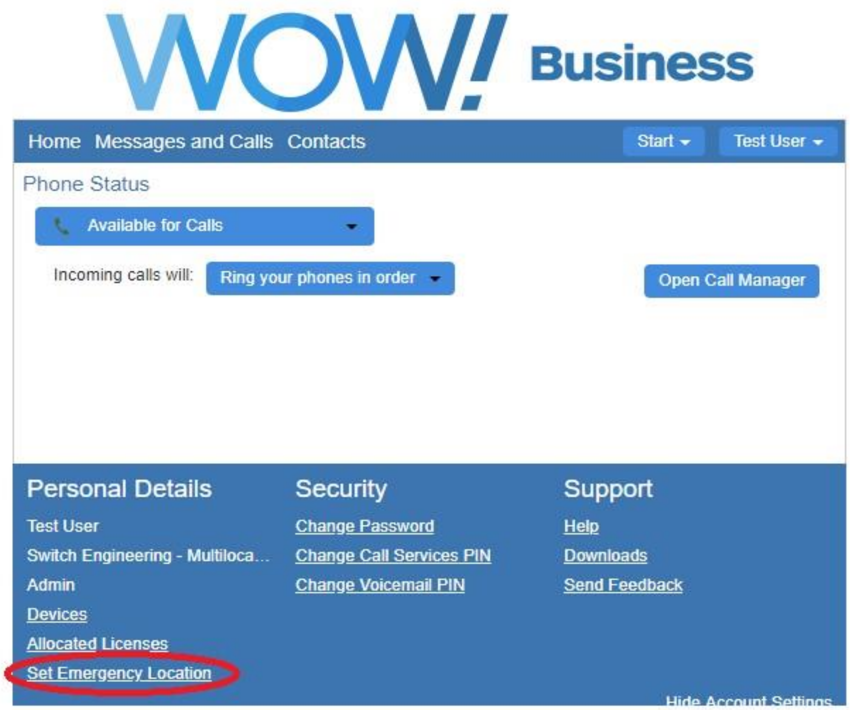
911 Considerations

We take our responsibility to set up 911 correctly with your Hosted VoIP services very seriously. On the initial installation of your services, the physical address of your main location is loaded into the 911 database. This will include descriptors such as building number, name, floor and room number, if applicable. This information is used to ensure that 911 calls are completed to the proper PSAP (Public Service Access Point) in accordance with FCC requirements.

If these phones or software clients are ever used in a location other than the original physical address, you, as the customer, are responsible for updating the 911 information for each of these phones. ***It is critical that this information is accurate at all times.*** Delays in providing the correct information or not routing the call to the appropriate PSAP could be disastrous in an emergency situation.

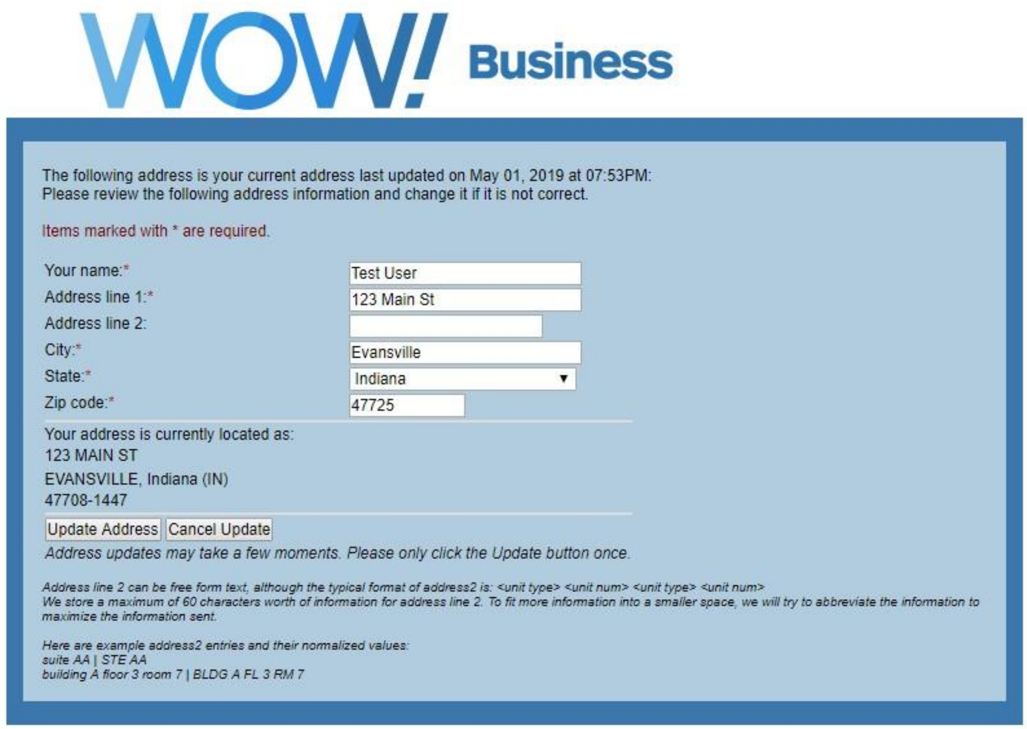
CommPortal 911 Instructions
You will find the following instructions in the Hosted VoIP CommPortal User Guide to maintain accurate 911 information at all times.

Go to Set Emergency Location for the specific user in the CommPortal.



Personal Details – Set Emergency Location

The Set Emergency Location link under Personal Details will allow you to change the address associated with your line for emergency (911) calls. For Address Line 1 enter the street address of your current location. Address Line 2 allows you to get more specific on your location at the street address. It is on this line, you can add your Building number, your floor number, your Suite or Office number or just a directional such as Office 24 in NW corner of the Building or West section of the building. Adding this information in Address Line 2 helps first responders to locate you more quickly. There is space for 60 characters in Address Line 2 and so please use Abbreviations when entering your location such as “BLD A, FL 10, STE 1000, OFC in NW CRNR”. Then please update your City, State, and Zip code.



Please contact us if you ever have any questions or concerns about setting up your 911 services. It is critical that this information is accurate at all times. During an emergency is not the time to wonder if it is correct.

Additional Support

Additional user guides and tutorials for your WOW! Business services can be found at https://www.wowforbusiness.com.

If you need any additional technical support, please contact us at 1-855-940-4969.

We are here to provide any assistance you may need for your Hosted VoIP services. Thank you for being a valued WOW! customer.

Definition of Terms

- **Packet:** The units that make up a data transmission within a computer network.
- **Latency:** The travel time for data packets, i.e., how long it takes them to reach their destination.
- **Jitter:** A measurement of the difference in latency between data packets when they reach their destination. A low jitter value indicates consistent transmission and ideal voice quality. A high jitter value can mean some packets arrive out of order or fall outside of a packet buffer.
- **Packet loss:** This occurs when some packets intended for a certain destination do not arrive as expected. The user experience of this would be poor or lost audio during a conversation.
- **LAN:** The Local Area Network is present within a customer's business.
- **WAN**: The Wide Area Network (usually considered to be the internet). A router will often have settings pertaining to the 'WAN' port, these will be the external side of the router.
- **VLAN:** Virtual LAN – A method used when it is necessary to segregate traffic within a single physical network.
- **CPE:** Customer Premise Equipment – The catch all term used to describe any servicing equipment present at a customer's business.
- **Soft Client**: A software application that allows you to make phone calls without needing a hardware phone.
- **IP:** Internet Protocol – The specific address designated to a device on a computer network.
- **Port:** The location at which a device communicates on a specific IP. Networks will typically use ports to designate traffic for a certain application.
- **SIP-PS**: The 'sip provisioning system' used by WOW to configure customer devices.
- **SIP Adjacency / Registering Adjacency**: The address (IP and port) on WOW's SBC that listens for devices attempting to connect for WOW Voice service.
- **DNS**: Domain Name System, an internet protocol used (among other things) to look up domain names and translate them into IP addresses
- **FQDN**: Fully-qualified domain name, a full DNS name that includes all levels of the hierarchy, e.g., 'inevvl.hvoip.voice.wowway.com'
- **SIP:** Session Initiation Protocol – The communication method by which WOW! delivers HVoIP services over an Internet connection.
- **OTT:** Over the Top – The term used to describe a provider offering VoIP services over another provider's internet connection.
- **Bandwidth:** The amount of data transmitted on a computer network within a fixed period of time. For Internet services, this equates to the subscribed service package. It should be noted that the bandwidth available to a specific device on a network (e.g. WOW! HVoIP phones and/or soft clients) can be restricted due to the usage of other devices on the same network.
- **ISP:** Internet Service Provider – The company through which the customer obtains internet access.
- **ZTP**: Zero-touch provisioning, a system that allows phones to be auto-configured to reach out to WOW's provisioning service without needing warehouse or field techs to program the phone on initial boot up.