Voice-over-Internet Protocol 101

A White Paper on the Fundamentals of Successful VoIP Deployments for SMB's
What You Really Need to Know About Implementing Hosted VoIP in Your Business

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Voice over Internet Protocol (VoIP) is a technology that has made rapid advancements over the past 10 years, allowing traditional phone conversations to be converted into IP packets (similar to an email or any other data transmission) and transported over properly designed IP networks, such as a customer private LAN, the public Internet (WAN), or a secure and managed MPLS Network (WAN). According to industry research, VoIP penetration among U.S. businesses alone will reach 79 percent by 2013, compared to 42 percent at the end of 2009.

Driving reasons for the implementation of hosted VoIP technology generally revolve around enhanced features and functionality, improved productivity, scalability and business solutions, as well as up-front capital reduction and long term TCO advantages (better, faster and cheaper). Some specific reasons for considering hosted VoIP include, but are not limited to, the following:

1. Newest Technology Constantly Upgraded with Economies of Scale on Hosted Platforms
2. Scalability – Up and Down – "Pay-as-you-Grow" Model
3. Free domestic long distance calling and highly discounted international calling
4. 3 or 4 digit dialing and intercom between offices and all company extensions worldwide
5. Centralized System Administration, Call Recording and Integrated Reporting/Monitoring
6. Virtual Local Market Phone Numbers giving a National Presence and reducing Long Distance Calls
7. Powerful Workgroup, Call Center and ACD Functionality (premise & remote workers)
8. Removal of all geographic and location barriers for remote offices and/or a distributed workforces.
9. Disaster recovery and business continuity benefits since the phone system resides in the cloud
10. Support issues addressed more promptly remotely by cutting down on truck rolls and wait time for a service technician to come on-site.
11. Productivity gains due to flexible architecture (API's), enhanced feature set, integration with other software (i.e. Outlook, Salesforce.com and others), and other versatility.
12. System can be administered by staff with basic IT skills and training, usually an office manager, etc.
13. Total Cost of Ownership (TCO) of 15% to 40% *less* than a traditional or premise-based phone system becomes an operating expense as opposed to a capital investment requiring depreciation. A typical business can experience a full ROI of their VoIP implementation in less than one or two years, if done properly from the onset.

NETWORK REQUIREMENTS

Below are the basic requirements your IP network (LAN and WAN) must be able to meet in order to be a candidate for VoIP, contrasted with traditional data network traffic, where the focus is on response time and throughput:

Broadband Network Access (WAN) Speed/Bandwidth

Bandwidth is the rate of data transfer which your Internet Service Provider (ISP) enables. There is a misconception that if you have any kind of "broadband" internet, your VoIP calls will sound perfect. While this does play a role, it is also important to realize that all of the other components outlined in this document are *equally* important in making for good call quality, and ultimately, the overall success of your VoIP deployment.

Bandwidth should be allocated to provide sufficient capacity for the maximum number of simultaneous calls at any given time (peak usage). Each concurrent VoIP call should have **dedicated bandwidth** available of 75 Kbps for the most commonly used G.711 "toll quality" codec (no compression) and only 8 Kbps for the less used G.729 "near toll quality" (compressed) codec. The growing deployment of the newer G.722 codec for high definition VoIP phones ("HD Voice") requires the same 75 Kbps of bandwidth as G.711.

You should use one of the multiple free online VoIP Speed Test tools to verify your bandwidth.

General Assessment of Network Bandwidth Options:

Business Quality Internet Options (Recommended)

1. MPLS: Private secured network with QoS and SLA, but can be expensive for smaller companies.
2. T1/DS3: Perfect bandwidth and functionality to support VoIP and QoS (Copper and Fiber options available).
3. FCC Licensed Wireless: Provided that it offers QoS and an SLA similar to MPLS.

Consumer Quality Internet Options (Non Mission Critical)

1. Cable: Depending on speed and ISP, can work well, but does NOT support QoS.
2. DSL: Generally works for 1-2 users maximum, but does NOT support QoS.
3. *Unlicensed* Wireless: Generally not recommended for VoIP .
4. Satellite: Not an option for VoIP due to inherent latency of up to 900 ms.

Important Note: Call quality can never be guaranteed with the use of the public internet, due to multiple carriers, hops and network routes using a "best efforts" method. However, many companies still successfully use the public internet for hosted VoIP, due to the cost and other benefits more than outweighing the occasional negatives. With this said, it is certainly ideal for customers to utilize the same network carriers at all sites as their chosen VoIP host provider to reduce network hops. The ongoing proliferation of "more bandwidth for the dollar" will continue to make this configuration even more reliable and efficient in the future, but in no way resolves the problem long term. As data-centric applications (such as HD Voice and Video) continue to explode, VoIP traffic will again be in contention with data applications, regardless of the amount of cheap "bulk" bandwidth thrown at the problem.

Enterprise and call center customers would also be advised to utilize managed network carriers who can offer a satisfactory Service Level Agreement (SLA), and integrated QoS, guaranteeing network performance. Some carriers like Qwest even post monthly core network performance statistics, but understand that these results generally do not apply to the "last mile" (local loops) where the traffic bottleneck typically occurs: http://66.77.32.148/statqwest/statistics.jsp. Additionally, network carriers should also be held accountable for not just an uptime SLA, but also a jitter and latency SLA. Further, network carriers should be held accountable for providing open visibility into these statistics to the client. Any network carrier providing high visibility and/or reporting on these crucial statistics on a real-time, daily, weekly, monthly etc. basis would be preferred, as problem resolution is dramatically shortened with this information.

Internal LAN Wiring

Although almost any kind of wiring infrastructure will support traditional analog and digital phones, the deployment of VoIP requires CAT 5/6 wiring that should be installed, tested and CERTIFIED by a professional. Commercial quality "Ethernet" wiring, connectors, terminations and patch panels should be utilized for each and every drop that is going to support a VoIP device. If your current wiring infrastructure is more than 5 years old and/or was not originally installed by a professional, it will likely not reliably support VoIP and will be highly susceptible to packet loss and intermittent call quality issues. Customers would be *strongly encouraged* to have a current wiring assessment performed by a licensed and qualified low voltage wiring contractor, with the proper testing equipment, who will provide a *detailed written report and certification* of testing on *each* port.

Network Switch Hardware on the LAN

Only a *managed* network switch should be used on the LAN, with QoS also enabled on the LAN (no hubs or unmanaged switches). They should also support V-LAN (Virtual LAN) configurations (separating traffic for VoIP phones from PC's and servers) and provide Power-over-Ethernet (PoE) to save from having to provide an additional power supply and transformer for each individual VoIP phone (tip: verify PoE on ALL ports; some switches only have 50% with PoE). Further, switch port mirroring and SNMP functionality would be recommended. Some suggested switch options are as follows:

Cisco SRW2024P 24-Port Gigabit Switch
D-Link DES-1228P 24-Port PoE Switch
AdTran NetVanta Model 1234 24-Port PoE Switch

Network Firewalls, Routers and Quality of Service (QoS)

Ideally, the VoIP phones would be placed behind a properly configured "SIP-Aware" firewall, but this will take the specialized knowledge of a professional who is familiar with your network to open the necessary ports on the firewall or router to allow traffic to/from the VoIP phones (get specific ports from your VoIP provider). Try to avoid NAT configurations (particularly double-NAT) for VoIP phones whenever possible. Consider using a router with FXO/POTS ports for analog line backup in the event your WAN connectivity fails.

Depending upon your configuration, you may need to implement port forwarding on your router. If you are using a shared network for both VoIP phones and computers, you are also well advised to enable "Quality of Service" (QoS) to prioritize mission critical VoIP packets over other network traffic. A preferred QoS method is to mark VoIP packets with the DiffServ setting for Expedited Flow (EF). Also, consider using Weighted Fair Queuing (WFQ), which raises the priority of low volume traffic. It should also be noted that once your VoIP traffic has passed your firewall or router and enters your WAN environment (internet, MPLS, etc.), the QoS tagging extensions will generally be *discarded* and not honored by your network carrier, unless the carrier has previously agreed to honor the QoS attributes. Generally, there is no QoS availability on the "public" internet like can be optionally provided by a single network carrier or MPLS provider. Customers should ensure that their QoS-enabled VoIP packets are honored intact as far into the network as possible, if not the entire call path.

On a shared network, a large computer download or streaming audio/video application can greatly hinder the VoIP call quality. QoS may be required at the host location and all remote locations, depending upon the configuration and bandwidth usage. QoS is an integral part of VoIP and without it, your call quality may suffer since IP bandwidth is dynamic. Please ask your VoIP provider for a list of specifically approved routers, but here is listing of several devices which generally work well for VoIP:

Edgemarc Appliances (Combined Router, Firewall & Traffic Shaper)
FortiGate 60B
Cisco ASA Firewall

For smaller installations, and remote VoIP extensions:

Linksys RV082, RV016, RV042
Linksys WRV210

IP Addresses / DHCP Server

Either static or DHCP IP addressing can be used for VoIP extensions. Phones are generally shipped to customer sites pre-configured for the use of DHCP, so the customer will need to have the technical expertise to individually change each VoIP phone to static IP addresses, if such a configuration is desired. Obviously, a DHCP server is required and the customer should assign an IP range or subnet which does not conflict with any other devices or computers on the network.

Latency

When you make a call using VoIP, that data is broken up into little packets and dispersed through certain channels of the network (just like an email). Latency is defined as the time between the moment a voice packet is transmitted and the moment it reaches its destination. Eventually, that call will reach the person on the other end. Latency basically refers to the geographic distance that call data must travel to reach your hosted VoIP provider, desired call recipient or other hop-off location. Latency through network delay finally leads to echo. It is generally caused by slow network links due to multiple hops or other delays. MPLS often helps alleviate latency issues as typical routing processes are circumvented with a much faster routing process (LDP). Try to choose providers that have worked to eliminate latency from their networks.

In order to insure good call quality, the network connection (LAN/WAN) should have latency (ping times) of less than 100 ms (150 ms maximum) to/from the host. One-way delay = propagation delay + transport delay + packetization delay + jitter buffer delay. It is highly suggested that customers utilize the same network provider(s) as the host VoIP carrier, both at any main office location(s) and remote sites (SOHO offices).

Jitter

In VoIP and data networks, jitter refers to the delta between the lowest measured latency and the highest measured latency. For example, if you run a ping test for an hour between a server and an endpoint, with the lowest recorded latency being 35 ms, and the highest being 60 ms, the jitter would be the difference of 25 ms. On the LAN, jitter should be 0 (or close to 0), but on a WAN it can be an issue requiring the use of a “jitter buffer,” which anticipates a certain amount of jitter and then corrects for it. Jitter can occur due to power surges, bandwidth congestion, or other irregularities in the system. The target is for Jitter of less than 60 ms on your network. If your hosted VoIP service provider has not perfected their system, there is a good chance you could be subject to jitter. MPLS often helps alleviate jitter issues as typical routing processes are circumvented with a much faster routing process (LDP).

Packet Loss

Packet loss occurs when a network connection becomes overloaded with data or traffic, or poor quality infrastructure wiring. Packet loss can be recognized during a VoIP call. It sounds like an echo, similar to having a conversation in a big empty room. Your LAN/WAN network connection and/or VoIP provider should not allow for more than 1% of packet loss, and obviously the less the better. If you are experiencing packet loss, consider cutting down on network tasks which overload your VoIP service and/or having your internal network wiring checked.

OTHER CONSIDERATIONS

3rd Party VoIP Network Providers (Carriers)

Whether you choose a more traditional premise-based IP-PBX or a Hosted VoIP provider, if you are going to have a quality experience (QoE) with VoIP, you must insure excellent network infrastructure ***from all parties***. This happens as a result of the investment in equipment, software, personnel, and level of maintenance provided by your 3rd party carriers. This also depends on the quality of the actual VoIP phones you are using; as with most things, you get what you pay for and the cheapest phones may end up costing you the most. There are many VoIP service providers in the marketplace who do not invest the amount of money necessary to offer business quality VoIP. Out of 500+ current providers, on average 20% cease operations each year because of undercapitalization or poor service, leaving their customers in a bad position.

Network Assessment

The importance of conducting a thorough network assessment, in advance, and working with a highly specialized VoIP professional cannot be over emphasized, and will play a direct role in the success of the deployment. Regardless of what some may purport, there is no substitute for an on-site network assessment in determining the viability of VoIP on your LAN/WAN. In many cases, special monitoring equipment will be connected to your network for multiple days in order to conduct an accurate assessment. Costs for a professional network assessment generally range from $750 to $1,500 depending upon what is included and the complexity.

Most hosted VoIP providers will accept the following 3 different types of network assessments and certifications:

1. Self Certification by Customer (Represents that they have the technical expertise to do so).
2. Certification by VoIP Provider, Installer or Dealer (Represents that they have the technical expertise and knowledge of the customer's overall WAN/LAN network to do so).
3. Certification by 3rd Party IT Vendor or Consultant (Represents that they have the technical expertise and knowledge of the customer's overall WAN/LAN network to do so).

Conclusion

While not solely conclusive, the use of a specialized online VoIP Speed Test can help you determine if your network (LAN/WAN) is ready to support VoIP.

Before undertaking any kind of sizeable hosted VoIP deployment, it is certainly advisable to "try before you buy" to make sure that the technology will work well in your environment and to identify any system limitations or surprises in advance. One of the many benefits of hosted VoIP is the ability to run service in parallel to your existing PBX until the number porting occurs, and it is suggested customers do this for a minimum of 30 days. Most carriers offer low cost paid "pilot programs" to evaluate their service, and this nominal investment will likely pay for itself many times over, as well as provide additional education on VoIP and the specific platform.

Information contained herein is general and not provider specific. Please consult with your chosen VoIP provider or one of our experienced technical consultants for the network details regarding VoIP *prior* to your implementation. It is highly suggested that customers include their network carrier(s) and/or WAN provider(s), in most, if not all, discussions regarding VoIP deployments, as your success is critically dependant on them. Unless expressly stated otherwise, generally the customer is fully responsible for providing, maintaining and troubleshooting the necessary network infrastructure outlined above to support VoIP, including but not limited to, LAN/WAN data circuits, routers, switches, firewalls, packet shapers, network interface cards, and all similar network equipment or other devices.

ADDITIONAL RESOURCES

VoIP 101 Training & Certification
VoIP for Dummies Book (XO)
Checklist of VoIP Design (NetIQ)
VoIP Mechanic
VoIP Certified Networks

TECHionary.com
10 Steps for VoIP (XO)
VoIP Network Assessments (NetIQ)
PCG Telecom Consulting Group, Inc.
Network Assessments (Imagine)

For Additional Information, Please Contact:

PCG
Telecom & Data Network
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VoIP Best Practices Table for LAN/WAN Network Configurations

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WIDE AREA NETWORK (WAN)

BEST	BETTER	GOOD	FAIR	POOR
Dedicated & Managed MPLS Connection for VoIP Phones	Dedicated T1, DS3, Fiber or FCC Licensed Wireless Internet Connection for VoIP Phones	Shared T1, DS3 or Fiber Internet Connection for VoIP Phones & Computers	Shared DSL or Cable Internet Connection for VoIP Phones & Computers	Satellite, Unlicensed Wireless or Consumer DSL Connection for VoIP Phones & Computers

LOCAL AREA NETWORK (LAN)

	BEST	BETTER	FAIR	POOR
ROUTER / FIREWALL	SIP Aware QoS Enabled Router with FXO Analog POTS Ports for Backup (Ex. Cisco, AdTran or Edgemarc)	SIP Aware QoS Enabled Router (Ex. Fortigate, Cisco, AdTran or Edgemarc)	Provider Uncertified or SIP Un-Aware Router	Inexpensive SIP Un-Aware Router (Ex. Linksys, Actiontec, etc.)
SWITCH	Managed Network Switch with Power over Ethernet (PoE), V-LAN Configured, and QoS Enabled on the LAN (Ex. Cisco, D-Link	Managed Network Switch with V-LAN Configured and QoS on the LAN, but not Power over Ethernet (PoE)	Managed Network Switch without V-LAN, Power over Ethernet (PoE) or QoS on the LAN	Unmanaged Network Switch or Hub without V-LAN, Power over Ethernet (PoE) or QoS on the LAN
WIRING	Dual Certified CAT5/6 Wiring Jacks to Separate VoIP and Computers	Single Shared Certified CAT5/6 Wiring Jacks for VoIP and Computers	Single Shared CAT5+ Wiring Jacks for VoIP and Computers	Single Uncertified CAT3 Wiring Jacks for VoIP and Computers

Note: The above information is general and not specific to any particular provider or platform. To a successful deployment of VoIP, customers are strongly advised to have a professional network assessment conducted, in advance, and to work with the IT staff and/or IT vendor to insure the proper network configuration.

VoIP Network Certification Form

www.VoIPCertifiedNetwork.com

Updated: March 28, 2010

Hosted VoIP has very specific LAN/WAN network requirements to support good quality VoIP calls, much of which is detailed in a document provided titled *"VoIP 101: A White Paper on the Fundamentals of Successful VoIP Deployments for SMB's."*

Customer and/or their IT vendor are solely responsible for providing, maintaining and troubleshooting the necessary network infrastructure required to support VoIP, including but not limited to, all LAN/WAN data circuits, routers, switches, firewalls, packet shapers, network interface cards, and all similar network equipment or other devices.

Choose your type of VoIP Network Certification:

1. ____ Self Certification by Customer (Represents that they have the technical expertise to do so).

2. ____ Certification by VoIP Provider, Installer or Dealer (Represents that they have the technical expertise and knowledge of the customer's overall WAN/LAN network to do so).

   Company Name: ____________________________

3. ____ Certification by 3rd Party IT Vendor or Consultant (Represents that they have the technical expertise and knowledge of the customer's overall WAN/LAN network to do so).

   Company Name: ____________________________

By signing below I/we agree that our LAN/WAN network(s) will meet the requirements to support VoIP, and we understand that troubleshooting provided by VoIP provider and/or VoIP vendor in this regard shall be billable service and support calls.

CUSTOMER NAME:	CERTIFIER NAME:
______________________________	______________________________
Signature of Customer	Signature of Certifier
Name:	Name:
Title:	Title:
Date:	Date: