hp procurve networking business

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technical brief

network infrastructure: getting started with VoIP

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introduction

In today's connected enterprise there is an intensifying dependence on communication networks for business success. From traditional voice systems and email, to instant messaging and video conferencing, next-generation communication technologies running on Internet Protocol (IP)/Ethernet networks are helping businesses turn mountains of data into valuable assets.

IP/Ethernet networks are fast becoming an attractive and viable utility for supporting high-quality voice in a converged network. Bringing disparate voice and data systems together into a multi-service network can create several compelling opportunities for achieving business results. Furthermore, by streamlining systems, Voice over Internet Protocol (VoIP) can help move businesses closer to anytime, anywhere communications.

The majority of VoIP implementations do not require substantial 'rip and replace' network overhauls or, for that matter, massively expensive network changes. What must change, however, is the way enterprises think of their network infrastructures. To deliver quality VoIP, enterprises must have a clear migration plan, the requirements in place for solid performance and a network architecture that can effectively handle quality of service (QoS) and bandwidth optimization. In addition, a network management strategy should include moving intelligence to the edge of the network. This approach effectively helps position an enterprise network for success by supporting all user types, making additions and changes easy to manage while enabling efficient use of network resources.

This paper is a guide for getting starting with VoIP. It examines strategies, tactics and basic network requirements for implementing VoIP. An enterprise should begin by looking at existing infrastructure capabilities and understanding basic network requirements and best practices for VoIP.

network optimization for VoIP

For optimal network performance and the best voice quality, network managers should evaluate their existing network and future communication needs. Enterprise telephony decisions should be driven by telephony requirements and business demands, and should not be restricted by the local area network (LAN) infrastructure. In other words, enterprise decision makers must assess and understand the critical set of telephony services and features required for their business when they prepare for and implement VoIP. Once business requirements are determined, the LAN can be designed, optimized and adapted to accommodate the VoIP solution. When making buying decisions, network managers should support industry standards to provide extra investment protection by ensuring interoperability between different vendors' products.

To begin a network assessment for VoIP, it is important to consider that voice traffic occurs in real time and is frequently impacted by delay, jitter and packet loss. To realize acceptable voice quality, HP ProCurve Networking suggests the following minimum requirements:
- Packet loss: ideal 1% between end points, however up to 3% will yield acceptable (toll quality) voice quality.
- Delay: 80-180ms delay between end points will yield acceptable (toll quality) voice quality.
- Jitter: 20ms or less will allow acceptable voice quality, however jitter is a function of network buffering and Type of Service (ToS).

bandwidth provisioning	Enterprises need a switched 10/100 network with a backbone connection of at least 100Mbps to support a multi-service network environment. If the embedded Web interface is reporting "high collision or drop rates," enterprises should consider increasing the wire speed or forming faster connections by combining multiple links using Fast EtherChannel or IEEE 802.3ad LACP. Network topology changes may also need to be made.

Enterprises must carefully analyze the network's wide area network (WAN) links if it will carry voice traffic. These links must be large enough to handle both data traffic and voice traffic needs simultaneously at a rate of 155Kbits/second for each active telephone conversation. This allows for uncompressed voice and includes protocol overhead.

end-to-end QoS

For end-to-end QoS, voice traffic must be prioritized higher than data traffic on each link along the communication path between IP telephones. It is recommended that network managers use IEEE 802.1p/Q-enabled switches to provide traffic class expediting. Routed networks should use DiffServ Code Points (DSCP), and networks with both switches and routers should use IEEE 802.1p/Q and DSCP as a best practice. IEEE 802.1p defines a priority field that can be used by LAN switches and works in tandem with 802.1Q. Specifically, the 802.1Q standard defines the operation of Virtual LAN (VLAN) bridges that permit the definition, operation and administration of VLAN topologies within a bridged LAN infrastructure. This reduces broadcasts, helps to increase bandwidth performance and improves security.

HP ProCurve switches provide 802.1p/Q capability and DiffServ in the LAN, so large file transfers or other data activity does not disrupt voice communications. HP network management tools can manage QoS policies in switches, routers, shapers, and end-hosts to provide end-to-end QoS on heterogeneous multi-vendor networks.

In the WAN, enterprises should establish a guarantee from their Internet service provider (ISP) for a maximum latency of 80ms between telephony end points and be sure to prioritize all voice traffic above data on all hops within the WAN.

VLANs and IP telephony

Along with priority tagging, VLANs can offer significant benefits in a multi-service network by providing a convenient way of isolating IP telephony equipment from the data traffic. When VLANs are deployed, excessive broadcast and multicast packets present in the normal data traffic will not disrupt IP telephony equipment.

The IEEE 802.1Q VLAN identification (VID) and 802.1p priority both reside in a tag inserted before the data in a packet. Many LAN switches can automatically perform tag insertion and removal for equipment that does not handle tagging. Packets for a given VLAN are always represented in the same way (i.e., tagged or untagged) on a given port. Therefore, both switches on a given inter-switch link must be configured in the same manner. To tag a packet, the LAN switch inserts four bytes just after the Ethernet source address (SA) field. The first two bytes are 0x8100, which is the standard Ethernet type value identifying the packet as tagged; the next two bytes hold the VID and the priority. The rest of the original packet data follows the tag. The cyclic redundancy check (CRC) is performed for error detection of the entire packet, including the tag (see Figure 1).

Normal Ethernet Packet

Destination Address (DA)	Source Address (SA)	Ethernet Type	Data	Cyclic Redundancy Check (CRC)

Tagged Ethernet Packet

Destination Address (DA)	Source Address (SA)	Tagged Ethertype (0x8100)	VID & priority	Ethernet Type	Data	Cyclic Redundancy Check (CRC)

Figure 1. Tagged and Untagged Ethernet Packets

A typical configuration is to create a voice VLAN (VVLAN) at each site. The VID associates a frame with a specific VLAN providing the information that switches need to process the frame across the network. The VID is a number from one through 4095, chosen for the VVLAN and used consistently throughout. The IEEE 802.1Q standard designates VID 1 as the "default" VLAN, and VID 4095 is reserved. Note that user priority and VLAN ID are independent of each other.

Assigning a different subnet number to the voice and data (default) VLANs is also recommended. If an enterprise has VoIP equipment at multiple sites, it should either bridge the VoIP traffic and use the same voice subnet number at all sites, or route it and assign additional subnet numbers if its VoIP equipment supports distribution across multiple subnets.

Tip: If an enterprise chooses to fully isolate voice and data networks by not routing between the voice and data VLANs, the outbound "telnet" command of HP ProCurve switches helps administer telephony equipment from anywhere on the voice or data VLANs. No routing is required. Network managers may telnet into the switch from the data VLAN, and from the switch console session, telnet out to telephony equipment on the voice VLAN. Switch management IP addresses must be configured on the voice and data VLANs in the HP ProCurve switch.

HP ProCurve switches also support the IEEE 802.1 GARP VLAN Registration Protocol (GVRP). This protocol allows devices to automatically advertise and register for membership in certain VLANs.

category 3 wiring

Wiring for many existing telephones is Category 3 (Cat-3), which is not compatible with 100Base-T. If an enterprise must use Cat-3 wiring, the HP ProCurve "Auto-10" feature allows configuration of a switch port to automatically negotiate full or half duplex, and reliably operate at 10Mbps on Cat-3 cabling. Also, it should be verified that cables are 100 meters in length or less.

outdoor wiring

Watching out for telephones placed in outbuildings or other situations where outdoor wiring may be in use is of extreme importance. There are electrical grounding and safety issues when using Ethernet on outdoor copper wiring. In these situations, a non-IP telephone should be used or a 100Base-FX fiber-optic connection installed.

power

IP telephones require power to operate. Several options for providing power exist today:

1. A separate AC transformer plugged into the wall next to the telephone is currently the common solution used.
2. Mitel offers a solution in which the AC transformer module has a battery backup and a single LAN wire connects to the telephone.
3. Standards are evolving to supply power over the Ethernet cable (IEEE 802.3af). HP ProCurve Networking will offer LAN switches capable of powering IP telephones later this year.
4. Another method of supplying power over the Ethernet cable is to use a powered patch panel. This device connects in line between any 10/100 LAN switch and IP telephone to provide power and can be deployed in a cost-effective manner such that it is wired to only those ports that need power.

911 emergency support

To provide communications in the event of a power outage, the network infrastructure must be connected to uninterruptible power supplies. As an intermediate step, designated emergency telephones connected to the local telephone company can be placed at strategic points in the building.

Most IP telephones can be easily moved from one LAN port to another without reprogramming. This can cause problems if the physical location of the telephone is different from the location reported to emergency personnel. To minimize this problem, a process should be established for keeping up-to-date data on physical locations of telephony equipment. HP ProCurve switches provide a MAC address-based security feature that can be used to lock down workstations or telephones to specific LAN ports, thereby preventing unauthorized movement of equipment.

existing analog equipment

Finally, enterprises must be sure the VoIP telephony solution chosen can be connected to existing analog equipment such as fax machines, modems and others at a reasonable cost. This will help increase the value of the investment.

VoIP with HP ProCurve Networking

Deployment of VoIP can be divided into three general stages:

1. Initial product and technology evaluation
2. Pilot projects
3. A phased deployment over time to each workgroup in the enterprise

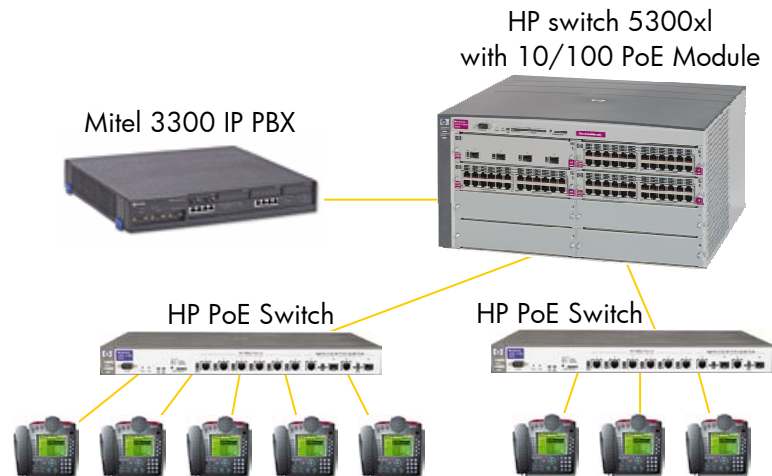## stage one – product evaluation

Figure 2. Stage 1 A VoIP Product Evaluation Network

In stage one (Figure 2), enterprises must create a stand-alone network solely for the purpose of learning how to configure the products and interconnect the VoIP equipment with existing telephony infrastructure. This allows network managers to evaluate voice quality in a controlled setting and to configure features such as call routing, billing, and automatic failover. The HP ProCurve Switch 2524m is an ideal low-cost switch with the features needed for IP telephony.

In the example evaluation network shown in Figure 2, ports connected to telephony equipment are configured to be members of the voice VLAN (e.g., VID 5). Ports connected to telephony equipment that does not do tagging should be configured as untagged members of the voice VLAN. Ports connected to workstations and servers are left as members of the default VLAN (VID 1). Each VLAN should belong to a unique IP subnet. If a second switch is connected to this network, the ports connecting the two switches must be configured for tagging on the voice VLAN, and no tagging on the default VLAN.

stage two – pilot project

During a pilot project example shown in Figure 3, the legacy private branch exchange (PBX) continues to serve the rest of the users, and a VoIP gateway transparently interconnects it to the IP telephones. This VoIP gateway is often pre-integrated into an IP PBX. Many IP PBXs have a full set of features and connectivity options, and can function as the sole PBX if need be.
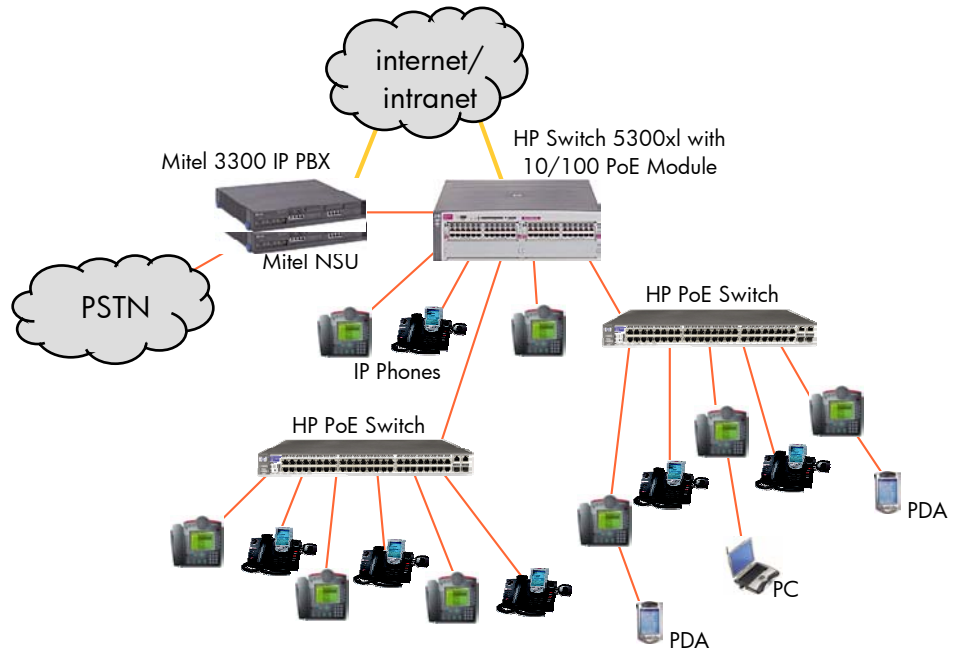

Figure 3. Stage 2 VoIP Pilot Project Network

**stage three -
extended
deployment**

As Figure 4 shows, VoIP should be deployed across the enterprise in phases. The key to successful deployment is proper bandwidth provisioning and end-to-end QoS. Voice traffic should be kept in a separate VLAN, and given priority over data traffic. If a WAN is in use, voice QoS must be maintained end-to-end across the WAN.
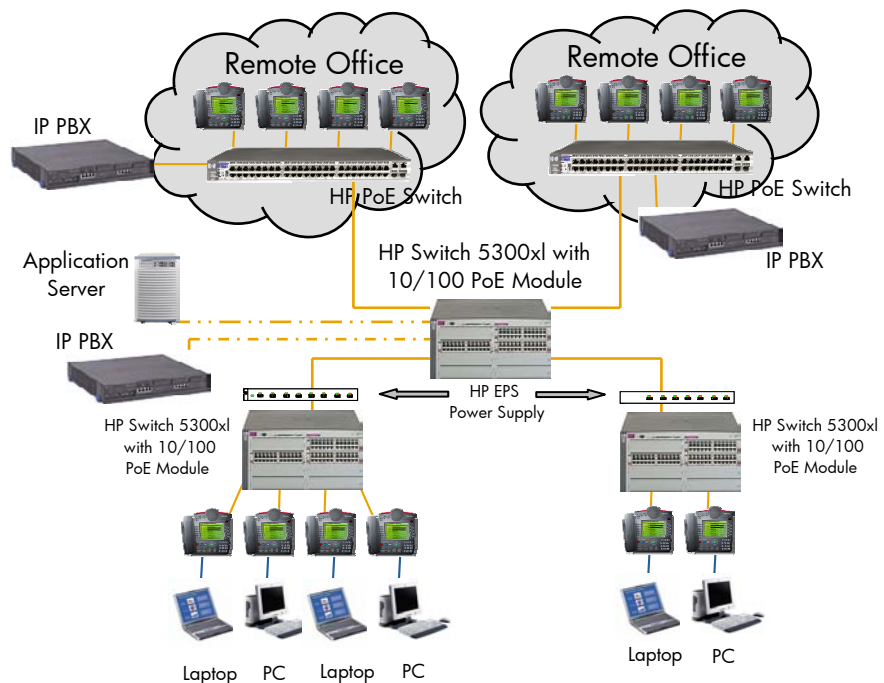

Figure 4. Stage 3 Extended VoIP Deployment Network

summary

While the momentum behind VoIP is still building, there are real advantages available today for enterprise companies that deploy the technology. For example, enterprises can realize reduced costs with one network by lowering overall support, equipment and carrier costs. In addition, a converged VoIP network is the foundation for deploying new-revenue generating applications. The technology, the multiple traffic types and users place new demands on the network infrastructure and all need to be managed appropriately to ensure a smooth transition to VoIP. Network managers should use industry best practices to avoid costly downtime and complete network overhauls. This includes doing an overall assessment of telephony business requirements and the existing network infrastructure. In addition, the deployment should be divided into three phases:

1. Initial product and technology evaluation
2. Pilot projects
3. A phased deployment over time to each workgroup in the enterprise

HP ProCurve Networking offers a variety of VoIP solutions for enterprise customers. HP ProCurve switches are interoperable and standards-based, so enterprises can choose the IP telephony equipment that best meets their needs. In addition, HP ProCurve provides the foundation for world-class IP telephony solutions by consistently delivering on its value proposition which includes the following key elements: high availability; affordability; security; ease-of-use and interoperability.

The company has a proven track record of invention and experience that allows it to deliver an adaptive and affordable infrastructure for today's and tomorrow's VoIP needs.

for more information

To learn more about HP ProCurve Networking solutions, contact your local HP sales representative or visit our Web site at: www.hp.com/go/hpprocurve.

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