IP Telephony Implementation Guide

February, 2001

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Introduction

Voice over IP (VoIP) is becoming a key technology consideration for many companies looking to, among other things, lower costs, take advantage of converged applications, or both.

The IP Telephony system is fundamentally an application on the network, and while it is true that “voice is data” once the analog waveform is digitized and packetized, it is a very specific type of data that has a very specific set of real-time requirements for the network infrastructure on which it rides. From a switching infrastructure perspective, Extreme Networks switches are exceptionally well enabled to support the addition of voice communication-*the* mission critical application-and have been designed that way from the very beginning.

Equally important to the capability of supporting the prioritized requirements of Voice over IP, Extreme switches are also able to simultaneously offer other guaranteed, high priority service to such applications as real time video, streaming audio and video, and various data applications like ERP with equal assurance-a true multi-service network infrastructure, as opposed to being simply a “one trick pony.”

While this guide is written toward IP telephony systems that use actual IP client based phones due to their more complex requirements from the infrastructure, Analog phone based Voice Switch systems are the other widely installed option for deploying IP telephony in the Enterprise campus and across it.

This guide intends to familiarize the reader with the Infrastructure issues surrounding an implementation of VoIP, specifically in the LAN/Campus, as well as discuss specific guidelines for configuration of Extreme switches to support IP Telephony (the terms, VoIP and IP Telephony are interchangeable and both will be used in this paper). Finishing out this guide will be the basic installation steps for installing the Cisco AVVID IP Phone System.

Switching Infrastructure Issues

Successful implementation of IP Telephony in the LAN/Campus, requires several things from the network infrastructure. While not a bandwidth intensive application, VoIP does require that the bandwidth be always available, and with Quality of Service that guarantees low latency (delay) and low jitter (delay variation).

<u>Switched Infrastructure</u>:

The first requirement is that the infrastructure be switched; both end to end and from Core to Desktop. IP Telephony will not work well on a network with shared Ethernet components. Today’s switched Ethernet networks running at 10/100 or 1000Mbsp typically have more than enough raw bandwidth to support VoIP, and these link speeds combined with the requisite QoS discussed below, it is easy to be under the end to end Delay Budget of 150Msecs, which is the maximum recommended for acceptable voice conversation.

Quality of Service:

Even though raw bandwidth capacity can be sufficiently high in a LAN/Campus network, there is still a need to provide a guaranteed level of packet throughput to VoIP for it to function well. QoS in this case is important for its ability to protect VoIP from the bursty, bandwidth hogging potential of many data applications and video. The three key component parts to Quos are Classification, or how the switch identifies traffic for various levels of Quos treatment, the h/w based queues, which are the physical buffers that allow for as many levels of distinct Quos, and the scheduling algorithm, which determines how the traffic that has been classified into a queue is actually given it's transmission opportunity at the egress port. These three components are all key to a robust servicing of not only VoIP on the LAN/Campus, but also need to be looked at for their ability to simultaneously provide QoS to other priority traffic on the multi-service network like Video and ERP data.

Classification-

The Real Time Protocol generally uses dynamically negotiated UDP port numbers, so this method of classification is not available to switches. Shoreline Communications is a notable exception to this with the consistent use of UDP port 5004. In the other cases however, there are a several other classification options available for providing QoS to VoIP traffic. For implementations with IP phones, the phones themselves may support both 802.1p and DiffServ explicit packet marking, and will generate frames/packets with the QoS level indicated as high. Another option for IP Phones, where the widely recommended configuration for IP Telephony support is as a separate VLAN, the Telephony VLAN itself can have appropriate QoS applied to it. Other options like IP Address or range classification can also be used, depending on the capability of the switches.

Queues-

Each hardware queue in a switch can support a unique scheduling configuration (described next), for once a packet has been classified into a queue, it will be First In/First Out of that queue. This means that for each type of traffic or application that will need differentiated treatment-QoS-from the network, there needs to be a separate queue that can be serviced with the desired parameters-bandwidth, priority, or round robin weight. If there are two queues, then only two levels of service can be met on the network.

Scheduling-

Again, the bandwidth requirement for VoIP is not especially large, as network applications go. A single IP phone call on an Ethernet network requires about 80Kbps (uncompressed 64Kbps voice + RTP/UDP/IP/Ethernet overhead), but that 80Kbps *must be guaranteed as available to that call whenever it is needed*. Quality of Service is the way to do this. There are two methods that can be used. One is to simply use **strict priority queuing**, where all voice packets would get priority transmit queuing treatment. Another method would be to set the **minimum bandwidth guarantee** for Voice such that all links have a preserved percentage representative of the numbers if IP Phones that could aggregate over that link. Simple Weighted Fair Queuing would not be adequate in many scenarios due to its lack of bandwidth granularity-it is fundamentally packet count based, not byte count based-for guaranteeing voice traffic, so it would be better to use strict priority if real bandwidth allocation is not an available option on the switch.

With eight <u>real</u> hardware queues per port, and <u>real</u> bandwidth allocation available bi-directionally, and the well known non-blocking, wire speed performance, Extreme switches are uniquely capable of supporting VoIP *and* other priority traffic types in the <u>real</u> multi-service network of today and tomorrow.

Physical deployment of IP Telephony Components:

There are two main system options for IP telephony systems, and these can be used in a hybrid deployment as well. IP Phone based systems have each phone Ethernet/IP connected to the network directly, while Voice Switch systems connect multiple, ordinary analog phones to a switch that provides connectivity and a gateway function to the Ethernet/IP network.

IP based Phones-

IP Phones can have a dedicated switch port connection devoted to the IP phone, but as this would require two drops to each desktop, a more likely scenario is to have an IP Phone share a switch port and cable with the desktop PC. This is often done with the IP phone having basically a three port switch built in, to that there is a 10/100 port connection from the phone to the switch, a 10/100 port connection that connects to the desktop PC, and the internal port that connects to the phone itself.

Analog Phones-

Analog phones must connect to an IP network via a voice switch, which is a gateway that will digitize and packetize the analog signals from 12 or 24 standard voice phones, and additionally provide the control and/or call signaling to the IP Telephony Gatekeeper, and other H.323 clients which can be IP Phones directly connected to the network, other gateways representing other analog attached phones, or gateways to the PBX or PSTN network.

Call Management-

Call Management is a key IP Telephony component that will be directly attached to the IP network. It can exist as a centralized Server with optional backup servers, or can be distributed throughout the network, depending on implementation. Additionally, an IP PBX, or an IP connection to a standard PBX would also offer direct attachments to the IP network and provide call management services.

Logical Deployment of IP Telephony Components:

The logical deployment will vary depending on both the implementation of VoIP, and also on the scale of the system. In systems that use IP Phones, each phone becomes an added Ethernet and IP station on the network-in essence a doubling of IP stations. Systems that use Voice Switches are less taxing to an existing IP network, as each switch can proxy multiple phones onto the IP network.

IP-based Phone System-

When deploying an IP based Phone system, there are three options for logical design that can be considered. The first would have each IP Phone handset added as a Client IP Address to the existing subnet. However this could require a doubling of assigned IP addresses, which may not be feasible. A second option is to create a new IP Subnet off the routed interface for the IP Telephony components, and add it multi-netted to the existing IP subnetted interface. The third option, and the one currently recommended by most manufacturers of IP phones, is to create a **Telephony VLAN**. The Telephony VLAN allows the easiest configuration and management options of directly attached IP phones, in addition to logical separation and prioritization of the VoIP traffic and transactions. It is the leading recommendation for IP Phone deployment made by vendors of such systems.

The Telephony VLAN is configured as an overlay across a network that can be either a routed network, a VLAN separated network or a hybrid. It will support only the IP Telephony application components and traffic, and is especially appropriate where a phone and a PC will share the same switch port from the closet, as it allows the use of 802.1Q to separate the voice and data traffic at the switch port. In this case, the phone typically provides 802.1Q tagging for VLAN separation. So even in the case where the PC Network Interface Card does not, it still works.

However the Telephony VLAN is enabled, it provides application isolation for QoS and administration, and also allows specific DHCP service to keep adds/moves/changes easy and cost effective within the Call Management zone. Using RFC 1918 recommendations for IP address range, the IP Telephony VLAN will also be it's own subnet, fully routable across L3 only links or to interconnect Telephony VLANs that have been separated for administrative or other reasons.

Voice Switch System-

Implementations that use analog phone gateways have easier configuration requirements from the network since multiple phones are represented by a single IP entity (the voice switch). Because of this, the additional required IP address space is much less of an issue since a set of voice switches can be added to existing subnets, regardless of subnet configuration, and configured to switch and/or route call between each other.

Also true for voice switch systems is that standard analog phones can be purchased and connected-keeping the complexity and cost of the handset low, while allowing a deployment of IP Telephony over existing voice grade cable.

In Line DTE power for IP Phone handset:

In line power is another issue specific to IP phone handsets, since analog phones connected to a voice switch receive power from the voice switch just as they would from a traditional PBX.

Many IP phones have or plan to have some kind of support for receiving power directly from the desktop drop cable that connects it back to the wiring closet. There is a standards effort underway as part of 802.3- specifically 802.3af, which seeks to finalize the technology currently by 2002.

Power can be “added” to the data cable in two ways. One is to have the power added at the switch port, and the other is to add power “mid-span” with a powered patch panel. Additionally, there are two supported options for delivering a low –48vDC current down the same cable that delivers Ethernet data. One uses pairs 45 and 78, while the other uses the actual 10/100BaseT data pairs of 12 and 36. The standard plans support for both, with the DTE (phone, Wireless Access Point or other Data Terminal at the end) “sensing” which method and adjusting itself to work either way.

Also part of the standard is the ability of the power-inserting device to “discover” or determine whether or not the attached equipment can and wants in-line power. Otherwise, the power-inserting device could potentially damage the NIC of anybody that connected to the “wrong” RJ45 jack.

Extreme Implementation Guidelines for VoIP

These guidelines are based on a collective of best practices proposed by of the major VoIP system vendors. Where Extreme switches in the infrastructure differentiate themselves is in their superior ability to very flexibly classify the voice traffic, and to guarantee the minimum bandwidth settings AND in their ability to offer additional guaranteed QoS to other traffic classifications on the network, an ability that provides true multi-service support.

Of course, as with any mission critical network application, the need for redundant systems and resilient protocols is a given. Such is the case especially with deploying VoIP, however those more general topics will not be addressed specifically in this paper.

IP Phones- The IP Telephony VLAN:

While Extreme switches support all of the logical configuration options, both in terms of pure mechanics and in terms of voice traffic classification and queuing, the general consensus best practice for deploying IP phone based telephony in the LAN/Campus today is via a separate IP Telephony VLAN.

So in terms of this VLAN support, there are a few capabilities that are especially useful in IP telephony deployments:

Tagged and Untagged ports in a Tagged VLAN-

The IP Telephony VLAN should be a tagged VLAN, with a specific 802.1Q VLAN ID that is selected for the IP Phones. If the IP phone and a desktop PC are sharing the switch port, the specific port should also be a member a different VLAN appropriate for the PC. This allows both the tagged IP Phone frames and the (probably) un-tagged PC frames to share the physical switch port, but achieve VLAN separation within the switch. Also, some IP Telephony devices may not support 802.1Q tagging, but would still need to be untagged-port members of the tagged VLAN.

IMPORTANT: If the same physical switch port is going to be shared by the IP Phone using TAGGED frames and a Desktop PC using UNTAGGED frames (either does not support the 802.1Q standard or is not configured to use tagging), the switch port MUST be configured as UNTAGGED or the PC data frames will be discarded by the switch.

Quality of Service:

Quality of Service for the IP Telephony traffic queue can either be guaranteed using strict priority, or by setting a sufficient Minimum bandwidth percentage to ensure that there is always enough of a given link to accommodate the 80Kbps per call. This is not a lot of bandwidth, even at the aggregation points, but still needs to be understood.

Strict Priority Setting-

If there are no other stringent QoS requirements on the network, in support of say real time video or other similar applications, or if the Voice load on the network is well understood (and scoped), simple Strict Priority queue scheduling can be used to ensure that Voice packets are always preferentially transmitted at the output port.

Minimum Bandwidth Setting-

A Summit 48 with 48 IP Phones needs to have a min set at 4Mbps of downlink for 48 simultaneous calls; 200 IP Phones connected to an Alpine, would need 16Mbps of downlink set as a min bandwidth parameter. This calculation of potential peak bandwidth requirement should be done at least roughly at the edge and core. Realizing that there is an actually low probability that all of the phones will be involved in calls simultaneous is where the beauty of the MIN bandwidth parameter comes in. While meeting the MIN bandwidth guarantee is the number one priority, in the absence of Voice packets in the queue, the bandwidth is freely available to other applications that may need it.

With 8 dedicated hardware queues and full bandwidth allocation of MIN and MAX parameters, the Extreme switch is of course supremely qualified to support IP Telephony. What really sets it apart as an infrastructure switch is the simultaneous ability to continue to service various other high priority application traffic types without compromising the degree of quality for any of them. This is true multi-service networking.

Read and Write capability for 802.1p and DiffServ code point-

For IP Telephony components that do not support 802.1p or DiffServ, the Extreme switch is a perfect enabler. The switch can be configured to actually add the desired .1p or code point value to the frame or packet as it leaves the switch, so that other switches downstream that would otherwise not be able to differentiate the voice traffic are now able to preserve the required “end to end” chain of QoS. Again, in the case of analog phones that connect to the network via a voice switch, the QoS can be more easily and manageably mapped by IP address of the voice switch.

In Line DTE Power:

Again, only an issue for the IP Phone based system. Extreme has tested interoperability with the two Powered Patch Panel vendors in the market today. One is Cisco, whose powered patch panel is proprietary in it's support for only Cisco IP Phones, and the other is PowerDsine, which OEMs to many of the others and sells it's own labeled version as well. For now, we would use one of these two options for providing in line DTE power. Given the early nature of the standards activity, this might be a good thing.

The ability to provide in line DTE power through an Extreme switch port is under technical investigation, as is a possible Extreme Powered Patch Panel, should the standard and or market become more solid.

Basic Installation steps for Cisco IP Phone System

Cisco Systems has evolved an IP Telephony system that it acquired with the purchase of a company called Selsius Communications. The system consists primarily of IP Phones, a Call Management Server. Cisco also makes IP to T1 gateway modules for the 6500 to connect to PBX or PSTN, but these components are easily, and preferably added to the network from many sources. Extreme has a Cisco IP Telephony system, and have tested with it. *The below notes and steps are designed for familiarization and to serve as a quick start, not to remove the need to read the Cisco IP Telephony system installation guide and release notes.*

[The system will require a DHCP Server, the Call Manager, and two or more Cisco IP Phones.}

Call Manager Setup

The Call Manager Server is a Compaq PC that comes with a clean HD. It has a single slot filled with a combo monitor and 10/100 ports, and built in mouse/keyboard connections.

1. The first step is to install the run-time NT OS and Call Manager software on the Compaq Server platform

   IMPORTANT: Read all instructions and release notes that accompany the Cisco IP Phones and Call Manager Server before installing the system.

2. The Call Manager itself should be given a meaningful Computer Name, which will be used for the Cisco Call Manager application and it's utilities. Once the Computer Name is selected during the installation, it cannot be changed except by reloading everything from scratch.

3. Assign an IP Address to the server. This Server will be the both the Call Manager AND the TFTP server that the phones will request configuration from.
4. The Call Manager utilities are all WEB based, and will make a "call" to the Call Manager name. So either network DNS can be used to resolve the call manager name to IP address, or two local files can be modified with the local DNS entry required.

   To do this, edit two files under C:\WINNT\System32\drivers\etc. The first file is <hosts> and the second file is LMHOSTS. Add the computer name and IP address match per the example in the file.

Configure Call Manager for IP Phone Registration and configuration

Using the GUI based Phone Configuration Utilility, set up the configuration for each phone.

1. Cisco Call Manager is available from Programs on the menu bar. Select Cisco Call Manager Administration and launch the Administration tool.

2. Configure Call Manager to provide Registration and call services to the phones.

There are two ways to set up the Call Manager to register phones. One way is to hard configure the IP phone MAC address into the Device Add field from the top menu bar on the Call Manager Administration utility. In this way, the Network Manager can hard assign phone extensions to exact phones. Another way is to have the Call Manager assign extensions serially to phones as they register. This method does not require the manual entry of MAC addresses, but outside of determining power-up order, there is also no way to specifically assign extensions initially. However, this is a "learn the first time" model, where once this initial registration happens, the Call Manager creates an entry tying the MAC address to the assigned extension, so that it will from then on, register that phone to that extension even if the phone is relocated within the VLAN.

DHCP Service Setup

Even though it appears that the Call Manager could serve as the DHCP server for the Phones, IT CAN NOT, unless there is a separate corporate DNS service that will resolve the Call Manager name to IP Address for the TFTP Request.

The IP Phones REQUIRE DHCP to start the configuration process. Except for the VLAN ID, which must be set on the phone, the Phone cannot be manually configured for operation, even though the menu will make it appear that you could. The DHCP can be scoped to provide the desired range of IP addresses.

If there is no standalone DNS service, the DHCP service must support OPTION 150- TFTP Server IP Address. This TFTP server IS the Call Manager, so will use the Call Manager IP Address. The Cisco IP phone will need this field to complete the registration to the Call Manager.

> *IMPORTANT: The DHCP service on the Cisco Call Manager does NOT support OPTION 150, so cannot be used for DHCP service if TFTP Server Address Option 150 is required, as in a network without DNS service and a DNS entry supporting the Call Manager..*

Powered Patch Panel setup

The powered patch panel is a very straightforward device. A 48 port powered patch panel will have 96 physical RJ45 jacks on it, in two rows of 48. One row will be labeled CLIENT PORTS and the other row will be labeled SWITCH PORTS. So, port 1 of the CLIENT ROW, connects to the desktop wiring, and port 1 of the SWITCH PORTS row connects to the switch port. The patch panel only senses and inserts in line DTE power, if there is a Cisco Phone connected to the other end. So, for port 1, connect one RJ45 to the switch port and one to the desktop distribution patch panel.

Switch

Create a Telephony VLAN as Tagged and assign the desired VLAN ID. The desktop ports should also have a VLAN (probably untagged) created for the PCs as a data VLAN.

Each physical port that connects to both an IP phone and a desktop PC, needs to be put into BOTH VLANs. The port should be an UNTAGGED port if there is an UNTAGGED PC connected.

QoS configuration can be either priority or Min bandwidth allocation, as described in the Extreme Implementation section.

IP Phone Setup

Power-

There are three options for providing power to the IP Phone handset. The 7960 IP Phone can accept In-Line DTE power from the desktop data drop, if it has an In-Line Power Patch panel installed Mid Span between the switch port and the desktop distribution patch panel. A second option is to purchase an external AC power supply plug that connects directly to each phone and powers at the desktop. The third option with the Cisco IP phones is to run power from both connections and use the In-Line DTE power as a backup.

VLAN ID (VID)-

Once the handset is powered up, it will go through a power on sequence.

Pressing the * button twice, and then the # button puts the phone into an Administrator state where Settings can be modified.

Press the SETTINGS button, and then arrow down through the menu presented on the screen to VLAN ID and enter the desired VID for the Telephony VLAN.

Validate the new setting

Save the configuration

Now the phone has the Telephony VLAN as it's permanent VLAN ID (802.1Q tag). Once connected to the switch port it will use DHCP for address and TFTP Server Address, and from there it will complete the registration process with the Call Manager. Once complete, the phone will display the assigned extension for that phone, and you are ready to call the other similarly started up phone-just dial.

The Cisco IP Phones generate CDP packets that a Catalyst switch could read and respond to with the VLAN ID assignment to the IP Phone-fully automatic configuration to the network out of the box.

With Extreme Networks switches as the infrastructure, currently, the only parameter that gets configured on the IP phone handsets is the VLAN ID for the Telephony VLAN. It could be possible to automate the VLAN ID assignment to the phone under Extremeware, and this is currently under consideration by Engineering.