Network VoIP for Corporate Environment Design

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Abstract: - This paper gives a global vision of the design of a VoIP (Voice over IP) network and its application in a corporate environment, studying technologies and present standards, and applying a solution given by Cisco Systems based on a Call Manager server. This paper discusses the different topologies that a system of these characteristics can support; also, the main configuration parameters are provided, and a step-by-step description about how to configure it.

Key-Words: - VoIP; Network campus topologies; Single/Multiple sites; Centralized/Distributed processing calls

1 Introduction
The main advantage of the VoIP is the economic saving. New data networks appeared in the last years offer a great speed at low enough cost. Accompanied this whereupon the electronics and computers have been reaching a considerable level of integration complexity and power, all it in an unthinkable cost for only 10 years. This power of the processors/DSPs/microcontrollers allows to implement the compression/decompression of the voice according to complex mathematical models in real time integrating of low easily burglary able cost. It is for that reason that the idea is made viable, integrating the traffic of inner voice to the medium of a great company in its data network, since its cost will be far below to the cost of the calls made through operator of telecommunications/traditional telephony.

2 Networks Structure
Figure 1 shows the general structure of a network. In access layer, level 2 protocols will be used (switching) due to high commutation speed and to the fact that tasks carried out by layer 3 protocols are not necessary. Any type of function will be solved by creating VLANs (Virtual LAN) and using the protocols Spanning Tree (STP) and VLAN Trunk (VTP). If multiple VLANs are created, it would be possible to make an internal classification of traffic without too much cost either in delay or in processing. Figure 2 shows examples of how to create VLANs in the access layer. The use of different VLANs makes possible to separate data traffic from vocal traffic, since each one of them will travel in a different VLAN (VVID: Voice VLAN Identification). At the same time, the VLANs going towards distribution network can be also distributed so there will not be congested connections and some other almost empty.

Through VLAN technology, switch ports and their users connected in work groups logically defined can be put into groups like the following ones:

- Fellow workers in the same department.
- Production team.
- Different user groups who share the same network application or software.
In our case, it is necessary to create VLANs to differentiate voice service in the different corporation offices. This can be done in two different ways:

Creating static VLANs: The static VLANs are ports in a switch that are assigned to a VLAN statically. These ports maintain the configuration of the VLAN assigned until they are changed. This requires a very high maintenance cost.

Creating dynamic VLANs: The dynamic VLANs are switch ports that can automatically determine their VLAN tasks. Dynamic VLANs are based on MAC address, logical address or data packages protocol types. For it a specialized server would be used.

The VLANs used would be the dynamic ones. The reason for it is because they benefit mobility within the LAN network, although a VMPS server (VLAN Membership Policy Server) is needed.

The protocol used will be the Spanning Tree Protocol, which is a connection protocol that provides route redundancy, and at the same time avoids the creation of loops in commutation network. In an Ethernet network, there is only a single active way between two stations. If there are several ways, loops appear and this makes a switch send the same message to that machine through more than one port.

Using VLAN Trunk Protocol, proprietary protocol of Cisco, reduces the administration in a switches network, distributing the VLANs in all switches under the same domain. This way, it is not necessary to configure all the VLANs in all the switches.

For distribution layer, there are several options concerning redundancy:

- Switches redundant with two supervisors can be implemented, so we get redundancy in devices and switching modules.
- Single redundancy in devices can be introduced.
- Implementing single redundancy only in supervisors’ modules.

In this type of network it is recommendable to use the first case exposed due to size and high availability importance. To have redundancy in all devices Hot Standby Routing Protocol (HSRP) should be activated. HSRP offers two possibilities: Method active-reserve: It is possible to have one of switches active and other passive. If the first one fails, the second one would become active. Balanced method: Network’s load can be shared by the two switches.

Distribution network will work with level 3 protocols, therefore it would be necessary to define the following parameters: Routing Protocols: Open Shortest Path First (OSPF), Inner Enhanced Gateway Routing Protocol (EIGRP) or Intermediate System-to-intermediate System (ISIS) Protocol to obtain network convergence.

Define the most standard configuration possible for all equipment to improve previous protocols convergence and problems solutions.

Implement route aggregation towards network core to reduce the number of area codes sent by the routing protocols and an excessive growth of its routes tables.

Core layer will strictly hold a transit function. It will also be based on layer 3 protocols, reason why we will be able to apply the same performance guide that stops distribution network in relation to redundancy and protocols.

3 Quality of Service (QoS)

With QoS system implementation in a network, it will be possible to make preferred treatments on certain type of traffic. This is useful at moments of congestion in some point of the network, because during those moments and without QoS configuration, the equipment usually makes packages discarding without any type of criteria. Nevertheless we can equip with certain intelligence this discarding can be equipped with certain level of intelligence if QoS is implemented.

The main goal of this section is to assign maximum priority to voice, then to video, voice and video signalling and, finally, data. As data traffic usually is non continuous traffic and insensible to losses or delays and voice has different...
characteristics, it is necessary to apply qualities of service that differentiate one sort of traffic from the another ones. Typical values are shown in Figure 3.

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Fig. 3: Typical values of QoS

4 Network campus Topologies
The different topologies that Call Manager allows are the following ones:
- Single site Model.
- Multiple sites with centralized processing calls.
- Multiple sites with distributed processing calls.

4.1 Single site Model
This IP telephony model consists in locating a calls processing agent in the network (LAN or MAN) without the need of offering telephony services through IP WAN network.

This design needs: A single Call Manager or a single Call Managers cluster, with 30000 lines per cluster (7500 lines by Call Manager).

For external calls RTC would be used.

The Voice Mail system would be managed by Unity. And single codec for the calls wold be used, for example G.711 without compression.

4.2 Multiply site with call control centralised
This infrastructure consists in a unique calls processing agent that needs to provide telephony service through IP WAN, so there will be remote locations that will use the Call Manager located in the central site.

When implementing this infrastructure, it is important to remember the following points:

- To provide high availability
- To design a dial plan to allow communication between extensions and towards the RTC.
- To use G.711 codec by all the user terminals, because the processing is reduced remarkably without reducing the bandwidth in our internal network.
- The use of Media Gateway Control Protocol (MGCP) for calls towards the RTC if H.323 functionalities is not needed because it will reduce dial plan configuration. H.323 is necessary if a certain specific functionalities of the Signaling System 7 (SS7) is needed.
- To implement QoS and security mechanisms in the network.
The things to consider in this configurations type are:

- It is necessary to reduce to the maximum the retardations between the remote sites and the CallManager.
- We will be able to give service about to 30000 telephones.
- The telephones will have to support the way of operation SRST (Survivable Remote Site Telephony).
- We will have retardation and limitations bandwidth in IP WAN.
- We will have to install the IOS telephony service in routers of the remote offices.

4.3 Multiply site with call control distributed

In this model, there can be several independent sites, each one with its own processing call agent connected to IP WAN to transport traffic between sites and to the RTC for external calls or backup connections. The main characteristic that differentiates this model form the previous one is that it is not necessary to send control signalling between sites.

Gatekeeper is one of the key elements in this model, where the rest of elements and recommendations are similar to those of the previous one:

- It is the element that will be done the CAC
- It will be able to manage entrance-exit bandwidth.
- It is convenient to use an only codec type for communications between sites through the WAN in order to simplify the planning, because the H.323 standard does not include any layer 2 heads, IP, UDP, and RTP in the used bandwidth control.

5 Conclusions

This paper presents a solution based on VoIP for a dispersed environment corporate, with offices in different locations of a country or, even, in different countries. At the same time, you can see that it presents the possibility of expanding the business of the company, offering telephony service supported by this system, as it is not difficult grouping various corporations and / or towns with different needs using different dialing plans, implementing a correct call control, providing the system the necessary channels of communication, etc.

Today there is a migration of traditional telephony to IP telephony, moving it much faster than we think, and that it has traditionally been beyond the reach of small and medium enterprises, due to high initial investment involved. The solution presented
here reduces barriers to entry for small and medium enterprises in this technology, using the concentration of different companies and/or users on a single platform Call Manager, enabling its implementation. Also, a solution is to give voice service and data entities, located far from the pockets of population, which has more cost-effective deployment with a single voice and data communications unified, and use their IP networks better than the traditional way.

References: