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Analysis of QoS: Classification and queuing techniques in VoIP Network

A Final Project submitted by

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ABSTRACT

Voice application in a packet switched network, Voice over IP is rapidly becoming popular. Besides valuable service for users, its sensitivity for loss, latency and jitter brings network technologists' attention into it and they came up with the idea of implementing QoS on VoIP network as a solution. This paper addressed the analysis of Class Based Waited Fair Queue (CBWFQ) and Priority Queue (LLQ) techniques on the current best practice VoIP network which is implemented in MINT lab. From experiments, the contribution of these techniques to the performance of VOIP services is investigated. Analysis of the impact of this technique on Voice traffic in the network is performed using wireshark network monitoring tool. Review of scientific papers regarding QoS for VoIP and descriptive survey of related researches are included. Finally the performance of these QoS techniques in handling voice traffic and its limitation are discussed.

CHAPTER ONE

1.1 INTRODUCTION

The global evolution of the Internet and the wide spread growth of networks have made the Internet part of our everyday life. For these reasons, the demand on the various Internet applications has increased. This increase in demand has also produced a number of new applications that did not exist before. Internet users can now access a variety of different multimedia applications such as video conferencing, Voice over IP (VoIP), and Video on Demand (VoD). Currently providing good quality of Voice through packet switched network has been receiving an ever increasing attention.

1.2 Statement of the problem and its Justification

1.2.1 Voice over IP (VoIP)

The research community shares the same meaning of Voice over Internet Protocol. Voice over Internet Protocol (VoIP) refers to the transmission of voice using IP technologies over packet switched networks. It consists of a set of facilities and protocols for managing the transmission of voice packets using IP. Internet Telephony is one of the typical applications of VoIP (1).

Compared to traditional resource dedicated public switch network (PSTN), IP network is resource shared. Therefore, IP based VoIP applications are cost efficient. The other benefit of VIOP is easy implementation of innovative services. These reasons alone would be attractive to any business contemplating installation of the technology into their business [2]. Open protocol and layer architecture of IP permit that VoIP technologies can not only realize existing quality telephone service, but also provide more and much better services, such as telephone conference and whiteboard [1).

Beside the valuable benefit, Voice applications have different characteristics and requirements from those of traditional data applications [3]. Because they are innately real time, voice applications tolerate minimal delay in delivery of their packets. Additionally, they are intolerant of packet loss, out-of-order packets, and jitter. To effectively transport voice traffic over IP, mechanisms are required that ensure reliable conveyance of packets with low and controlled latency [2] [4].

1.2.2 Quality of Service (QOS)

QoS is becoming increasingly important as networks get more crowded while more sophisticated Internet applications and services get deployed [3]. It is the term used to define the ability of a network to provide different levels of service assurances to the various forms of traffic. It enables network administrators to assign certain traffic priority over others or actual levels of quality with respect to network bandwidth or end-to-end delay [6]. The goal of QoS is to provide guarantees on the ability of a network to deliver predictable results. Taking this into consideration, this study aims to implement and analyze the widely used QoS technique on the current best practice Voice over IP network design, and give any possible recommendations.

1.3 Literature review

Quality of Service (QoS) is an increasingly important area of research and development in computer networking. [6][4]It is especially important for the new generation of Internet applications such as video-on-demand and other consumer services. Here are a few reasons to deploy QoS in a network topology: To give priority to certain mission critical applications in the network; To maximize the use of the current network investment in infrastructure; Better performance for delay sensitive applications such as Voice and Video; And to respond for changes in network traffic flows

In this day and age of Gigabit Ethernet, wireless and Optical Networking, higher capacities of network bandwidth are readily available. More bandwidth does not, however, always guarantee a certain level of performance [7]. It may well be that the very protocols that cause the congestion in the first place will simply eat up the additional bandwidth, leading to the same congestion issues experienced before the bandwidth upgrade. A more judicious approach is to analyze the traffic flowing through the bottleneck, determining the importance of each protocol and application, and determine a strategy to prioritize the access to the bandwidth. Quality of Service (QoS) allows the network administrator to have control over bandwidth, latency, and jitter, and minimize packet loss within the network by prioritizing various protocols [8].

1.3.1 QOS mechanisms

Two distinctly different philosophies were developed to engineer preferential treatment for packets which require it. Early work used the "IntServ" philosophy of reserving network resources. In this model, applications used the Resource reservation protocol (RSVP) to request and reserve resources through a network [10]. The second and currently accepted approach is "DiffServ" or differentiated services. In the DiffServ model, packets are marked according to the type of service they need. In response to these markings, routers and switches use various queuing strategies to tailor performance to requirements. At the IP layer, differentiated services code point (DSCP) markings use the 6 bits in the IP packet header. At the MAC layer, VLAN IEEE 802.1Q and IEEE

802.1D can be used to carry essentially the same information. IntServ and Diffserv mechanisms use packet processing tasks such as traffic classification, queuing, shaping, policing and scheduling.

1.3.2 The importance of Quality of Service for Voice over IP

Traditionally packet switched networks did not require strict measures for QoS because the data wasn't multimedia and the end-user would not notice or be materially affected by latencies. But, as the use of internet has spread far beyond simple data transfer to intense multimedia applications, the need to address Quality of Service (QoS) issues has become more important. Both the enterprise and consumer markets are now beginning to demand data intensive, time-sensitive movement of things like audio and video around Internet [2].

Transmission of voice traffic has to meet stringent requirements on packet delay as it is an important factor that affects the quality of calls. The International Telecommunication Union (ITU) recommends that a one-way delay between 0 – 150 ms is acceptable in Recommendation G.114 [8] Studies also show that jitter can cause gaps in the play out of an audio stream in an audio-conference. Moreover, packet loss would introduce considerable impairment for voice signals. The voice over IP (VoIP) generally requires some guarantees in terms of delay jitter and packet loss [4].

1.3.3 Related Works

Implementing Quality of Service in VOIP network has been researcher's subject of interest as long as voice has been identified as a critical real time application in a packet switched network. A number of comprehensive studies have been published over the years as the VoIP has evolved from 1995 the first Internet Phone Software appeared – Vocaltec, [11] to today's multi provider commercial environment.

The following literatures have been conducted in the respect of QoS for VoIP. The studies focused on the aspect of design, implementation, and management aspect of QoS on the VoIP network.

1. Survey on QoS Management of VoIP [1]

The main purpose of the study was to address a survey on management mechanisms used for ensuring the Quality-of-Services for Voice-over-IP applications. To accomplish the study, they partition the system into two management planes; the data plane and control plane, and describe mechanisms in each plane respectively.

In data plane, they cover several important techniques including*:*

- Packet classifier: which determines the packet flow or class it belongs to.
- Buffer management: which is designed to absorb short term packet bursts, keep queues at or near maximum occupancy and cause unfair resource usage. Active queue management, such as Random Early Detection (RED) and Fair RED which drops packets before a queue becomes full, can avoid these problems.
- Scheduling policy: which is primarily to control queuing and delay bandwidth sharing. There are varieties of scheduling disciplines, such as First Come First Serve (FCFS), Static Priority (SP), Priority Queuing WFQ, Low Latency Queue (LLQ), Weighted Fair Queuing (WFQ) and Class-Based WFQ (CBWFQ), Earliest Deadline First (EDF).
- Loss recovery: to recover the original content of the lost packet.
- Error concealment: is to produce a replacement for a lost packet which is similar to the original lost packet.

In control plane, they describe admission control, resource provisioning, traffic engineering, and congestion management: They also discuss methodologies used in admission control in detail. Specifically, they examine parameter-based and measurement-based admission control algorithms, and their use in VoIP.

- Admission control and congestion management are method of restricting the total amount of traffic competing for the same resource. They introduced two main admission control approaches parameter based and measurement based. Parameter based is a traditional admission control scheme, which computes the worst case delay or packet loss, using the information of traffic profile of the existing and new flows, and then makes admission decision.

- Measurement-based admission control algorithms can be classified into two types, local load measurement -based and end-to-end load measurement-based. With local load-based measurement algorithms each router along the flow path measures its local load status and conducts admission control. End-to-end measurement based admission control algorithms rely on the end hosts or edge nodes to estimate the available network resources.

- Resource provisioning refers to the configuration of resources for applications in the network. The amount of provisioning depends on the previous aggregate user demand. This mechanism allocates network resources at the network configuration time.

Traffic Engineering is concerned with the performance optimization of traffic handling in operational networks. Its main focus is minimizing over-utilization of a particular portion of the network while the capacity is available elsewhere in the network.

Finally, they discussed limitations of current technologies and future research issues in providing QoS to VoIP applications as follows: While much progress has been made in VoIP related QoS mechanisms, several issues remain to be addressed in order to fully deploy the technology: for example, signaling protocols of QoS controlling need to be further studied and analyzed; multi-level VoIP services are yet to be considered. Some mechanisms in data plane and control plane, such as flow-based classifier and scheduler, signaling, admission control, bandwidth broker and inter-domain QoS provisioning require further refinements for a scalable and efficiency implementation of QoS guaranteed VoIP system.

2. Dynamic Class-based Control for Improving Perceived Voice Quality [4].
Discussed as the current Internet does not have any inherent support for Quality of Service (QoS) guarantees. However, the voice over IP (VoIP) generally requires some guarantees in terms of delay and packet loss. They explain as the current QOS queue mechanisms minimize the amount of delay and delay variation. However these solutions do not provide perceptual QoS accord with users' expectation very well. This occurs because delay is not related to perceptual voice quality in a simple way but also user characteristics, voice codec type and loss recovery technique. Regarding the packet loss issues, there are two main approaches: Forward Error Correction (FEC) and Packet Loss Concealment (PLC). on the other hand these techniques introduce additional delay and overhead bit rate. They refer recent studies as packet loss can exhibit temporal dependency or bursts, which degrade the effectiveness of the PLC techniques.

To overcome the above problems and technique limitations, they have designed a Dynamic Class-based Control (DCC) approach to address the QoS issues in VoIP networks. And their goal is to improve the perceptual voice quality without incurring additional data or computation overhead at the sender or receiver side. The DCC scheme is based on careful consideration of user characteristics, voice codec type, and relationships with delay and packet loss.

They classified Dynamic Class-based Control scheme (DCC) consists of mapping and dynamic priority transmission mechanisms.

- A mapping procedure determines the maximum tolerable packet loss rate and delay for each voice flows. The E-model is first adopted to characterize the requested perceptual voice quality. In the E-model, the perceptual voice quality is based on the rating scale R and R can be derived by quantifying the various transmission impairments including delay, loss, echo, loudness, and frequency response. Each of the impairment factors is mapped to a score and R is the summation of these scores. When the default values for some factors are assumed, the simplified E-model is interpreted as : $R = 94.2 - I_d - I_e$

  Note: the E-model they used is ITU, Recommendation G.107

- The dynamic priority transmission mechanism assures that each voice flows meets delay and packet loss constraints. It dynamically assigns the voice flows into classes with different transmission priorities according to the QoS Turple { R_{PLR}, R_{Delay}}. They derived the formula from:
  - Let Δ_{PLR} be the difference between actual PLR and maximum tolerable PLR,

- R_{PLR} is the ratio of Δ_{PLR} and maximum tolerable PLR.
- Let Δ_{Delay} be the difference between actual one-way delay and maximum tolerable one-way delay,
- R_{Delay} is the ratio of Δ_{Delay} and maximum tolerable delay.

As conclusion, they propose the Dynamic Class-based Control scheme (DCC) for VoIP applications. Simulation experiments show that the DCC scheme achieves more voice flows with PSTN equivalent voice quality for given network conditions, yielding better QoS satisfaction both for voice users and network providers. Add on, they differentiate the study in to two important respects from previous studies of QoS control schemes used for voice transportation in IP networks. First, the new design provides better user-perceived voice quality than traditional QOS control technique. Secondly, the DCC scheme controls both delay and data loss, while existing scheduling approaches are primarily used to control bandwidth sharing and queuing delay.

Finally, they note as their future work includes investigating DCC under more complicated network conditions. For example, one scenario being investigated is where more than one router adopts the DCC scheme. Then, they will consider the effect of various network parameters, such as link utilization, number of hops and link bandwidth, on voice delay and loss.

3. Design and Implementation of QoS-Provisioning System for Voice over IP [7]

To address the current QoS issues in VoIP network, the researchers come up with new design and implementation of a practical, QoS-provisioning VoIP system to provide the end-to-end QoS guarantees to

VoIP networks. As of their explanation, the existing VoIP systems aim to provide a certain degree of QoS to voice over IP networks. However, none of these systems can provide end-to-end QoS guarantees. They introduce the general architecture of the existing commercial VOIP system and why these systems cannot provide QOS guarantees as follows.

Utilization-based Call Admission Control (CAC) is one type of CAC mechanisms. Two kinds of Utilization-based CAC mechanisms, Site-Utilization-Based CAC (SU-CAC) and Link-Utilization-Based CAC (LU-CAC), have been adopted by some VoIP systems.

- Bandwidth preallocation: Since the bandwidth has been pre-allocated to the sites at the configuration time, links cannot be fully shared by dynamic calls and, accordingly, high network resource utilization cannot be achieved.

- LU-CAC mechanism: This mechanism needs resource reservation on individual links in a network. Such a resource reservation approach will introduce significant overhead to the core-routers and, hence, greatly comprise the overall network performance.

The following are their network design and network architecture strategies to resolve the above problem.

Design

- Integrating a new QoS provisioning system to enable both SU-CAC and LU-CAC to be well utilized and supported. With this system, the overhead of resource reservation at the core routers will be pushed to the agents in the QoS-provisioning system, which overcomes the weakness of the current VoIP system in applying the

LU-CAC mechanism. They will also address the issue of performing resource allocation to better support the SU-CAC mechanism. .

- Static-priority (SP) scheduling: it is class based and its overhead introduced to the data plane is small and independent from the flow population.

- Linear programming approach to optimize the resource allocation in the control plane, while still providing the end-to-end guarantees with SU-CAC mechanism.

Architecture:

In the architecture and its components of their network, they introduced their QoS-provisioning system integrating with the commercial VoIP system. The system consists of three kinds of components.

1. QoS Manager (QoSM): The QoSM implements three basic functions:
1. It provides user interface to control and monitor the components, which are in the same QoS domain. 2. It provides registration to the distributed agents and coordination among the distributed agents in the same QoS domain. 3. It cooperates with the peer QoSMs that belong to other QoS domains.

2. The Call Admission Control Agent (CACA consists of two modules: 1. Utilization Computation Module performs delay analysis and computes the maximum bandwidth utilization. 2. Admission Decision Making Module will make an admission decision for each incoming call request, based on the allocated bandwidth utilization by the utilization computation module and the currently consumed bandwidth.

3. Integration Component (IC): The IC integrates CACA into existing VoIP systems and provides call signaling processing modules to monitor and intercept call setup signaling from Gatekeeper or Call-Manager, withdraws the useful message and passes it to CACA, and executes call admission decision made by CACA.

After all the design and implementation work, they did performance evaluation test. For this test, they consider two metrics 1. The introduced overhead to the admission and the overall bandwidth utilization. Correspondingly, they choose two measurement metrics: admission latency and admission probability. Admission latency is used to measure the overhead of admission. Admission probability is the ratio of the number of admissions to the number of overall requests, which is a well known metric to measure the overall bandwidth utilization. The higher is the admission probability, the higher is the overall bandwidth utilization achieved.

For Admission latency test, they run a suite of experiments with their CACA and without their designed CACA. Due to the different design and implementation methodology of CACA for CallManager and Gatekeeper, they run two experiments for both cases. The experiments are run in the Internet2 Voice over IP Testbed in Texas A&M University.

For Admission probability simulation, they consider two different network topologies Internet2 backbone network and a campus network. Gatekeeper and CallManager are configured to perform call admission control in the Internet2 environment and in the campus environment respectively.

In final remark, they point out as their QoS provisioning system has been successfully realized in the tested environments. Recall that their system adopts a static priority scheduler. With such schedulers, low priority traffic has no impact on high priority traffic. They also mention in their new system, voice traffic is assigned the highest priority so that knowledge of other highest priority traffic only is needed. Furthermore, their utilization delay analysis makes the system independent of the dynamic distribution of traffic which has the same priority as their voice traffic.

Result of integration of their proposed QoS provisioning system with the existing VoIP systems practical applicability with: 1, Closed networks where all traffic is under control like enterprise networks and networks on ships, and space shuttles. 2. Semi open networks, like Internet2, where all highest priority traffic can be known, although cannot be controlled. 3. Open networks, like the Internet, where traffic cannot be controlled and is difficult to be predicted. Their approach can provide statistical guarantees as long as the traffic in the Internet can be modeled.

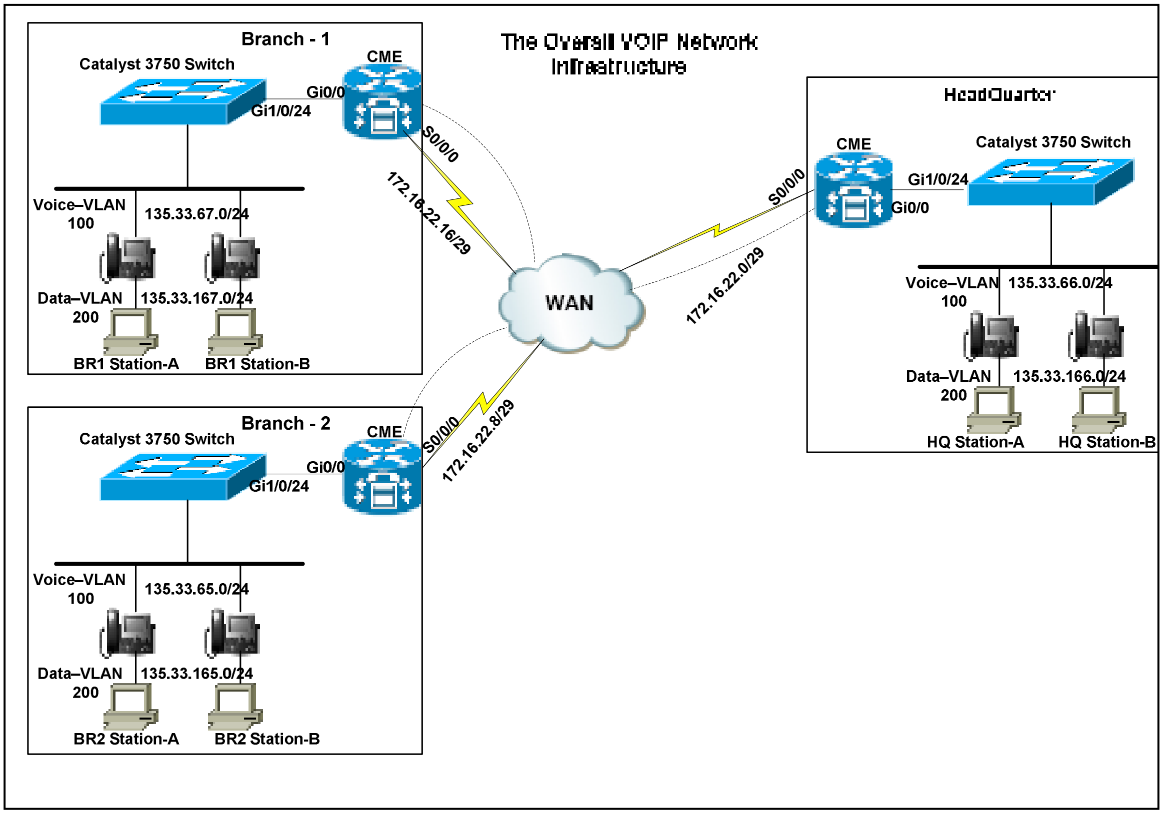
CHAPTER TWO

2.1 Design and Implementation

2.2 Network Topology

The very common commercial voice over IP network design is considered and physically implemented in the MINT lab. As illustrated in fig.2.1, HeadQuarter and branch offices are interconnected across the IP Wide area network (IP WAN). The network scenario in each location is symmetric. Figure2-1, illustrates the overall network topology and distributed call processing deployment of this work.

Fig2.1, The overall VoIP Network infrastructure



2.2.1 Network Components

The multisided Leased line WAN model with distributed call processing consists of multiple independent sites, the HeadQuarters, Branch-1 and Branch-2. Each single site uses Cisco Unified CallManager call processing agent (CME). The Cisco 2821 routers located at branch offices and HeadQuarters are performing three tasks. Edge routers to the WAN, Edge routers to their own site, and are work as Call manager and TFTP server for IP phones voice traffic process.

The Cisco 3750 catalyst switches represent the sites access layer connectivity. A single IP subnet per virtual LAN (VLAN) is assigned. This practice eliminates topological loops at Layer 2, thus avoiding temporary flow interruptions due to Spanning Tree convergence [12]. More over separate VLANs for voice and data devices at the access layer provide ease of management and simplified QoS configuration. To provide high-quality voice and to take advantage of the full voice feature set, access layer switches are configured for:

- 802.1Q trunking for proper treatment of Layer 2 CoS packet marking on ports with phones connected.

- Spanning Tree Protocol (STP) portFast is enabled to minimize convergence times and maximize fault tolerance at Layer 2 access ports. The phones and PCs, connected to these ports do not forward bridge protocol data units (BPDUs) that could affect STP operation. PortFast ensures that the phone or PC, when connected to the port, is able to begin receiving and transmitting traffic immediately without having to wait for STP to converge.

The 7941 IP phones Provide voice communication over the IP WAN or to their belonging branch site. They also support inline power and has an integrated 10/100 Ethernet switch for connecting a PC. All 7941 IP phones are using same VoIP codec G.729 and running the same Skinny Client Control Protocol (SCCP) signaling. The IP phones and PCs are accessing separate VLANs. For the purpose of link bandwidth congestion at the WAN and LAN edge interfaces, the PCs are used to generate and send traffic each other across the WAN, and to their belonging LAN PC.

2.3 Quality of service (QoS) Design

If the traffic needs to be prioritized on WAN circuits using some form of queuing technique like CB-WFQ and PQ is recommended. On this paper, CB-WFQ and PQ technique are implemented and analyzed.

2.3.1 Class Based Waited Fair Queuing (CBWFQ)

Class Based Waited Fair Queuing (CBWFQ) has all the benefits of WFQ, with the additional functionality of providing granular support for network administrator defined classes of traffic [15]. CB-WFQ enables network administrator to define what constitutes a class based on criteria that exceed the confines of flow, create a specific class for voice traffic and it enables the network administrator to specify the exact amount of bandwidth to be allocated per class of traffic.

With CB-WFQ, each class is associated with a separate queue. A specific minimum amount of guaranteed bandwidth can be allocated to the class as a percentage of the link, or in kbps. Other classes can share unused bandwidth in proportion to their assigned weights. When configuring CB-WFQ, bandwidth allocation does not necessarily mean the traffic belonging to a class experiences low delay.

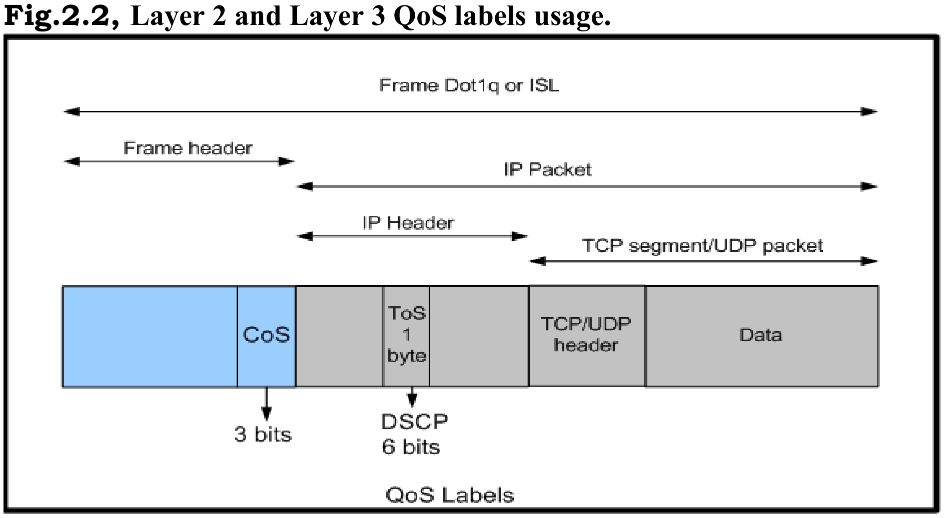
2.3.2 Priority Queue within CB-WFQ (Low Latency Queuing)

This queuing mechanism was developed to give absolute priority to voice traffic over all other traffic on an interface [12]. The LLQ feature brings to CB-WFQ the strict-priority queuing functionality of IP RTP Priority required for delay-sensitive, real-time traffic, such as voice. LLQ enables use of a strict PQ. Although it is possible to queue various types of traffic to a strict PQ, it is strongly recommended that to direct only voice traffic to this queue [16]. This mechanism has proven to be powerful and it gives voice packets the necessary priority, latency, and jitter required for good-quality voice.

2.4 Routers and switches QoS Design

QoS provides preferential treatment to certain types of traffic at the expense of others. It differentiates the traffic using QoS labels. The two most commonly used QoS labels in the Layer 3 IP header are the IP precedence field and the DSCP field. The QoS label in the Layer 2 frame header is called Class of Service (CoS)[12, 14].

Fig.2.2, Layer 2 and Layer 3 QoS labels usage.



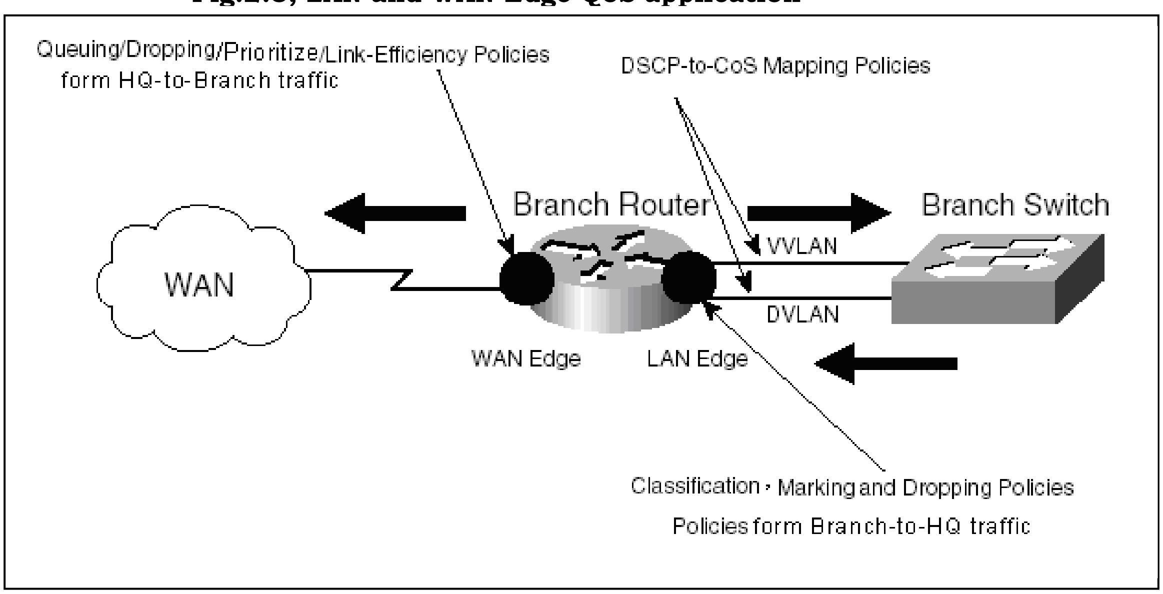
Unlike the routers, the QoS classification and marking act differently in Cisco Catalyst switches. In Cisco routers, can classify the packets using

Modular QoS Command line interface (MQC) either based on the incoming packet DSCP value or based on the access control list (ACL)[12,13,14]. This depends on whether the QoS label of the incoming packet is trusted or not. In the Cisco Catalyst Switches, frames can be classified either based on the incoming CoS/DSCP values or based on the ACL. On a well engineered network, care must be taken to separate edge and backbone functions to achieve the best QoS possible. For this work, QoS configuration design is implemented on the routers and switches as follows.

2.4.1 WAN-Edge and Branch Edge Routers QOS Design

In this paper, each of the routers on Head Quarter, Branch-1, and Branch-2 play two connectivity roles: as a WAN aggregator and a Branch router. QoS policies required on WAN aggregators include queuing, selective dropping, and link efficiency policies in the outbound direction of the WAN link. Fig.2.3 shows the QoS application on WAN and LAN edge.

Fig.2.3, LAN and WAN Edge QoS application



Traffic is assumed to be correctly classified and marked at Layer 3 before WAN aggregator ingress [12]. Traffic destined to the far end Branch or HQ might not be correctly marked on the branch access switches. These switches, which are usually lower-end switches, might or might not have the capabilities to classify by Layer 3 or 4 parameters and mark DSCP values for data applications. Therefore, classification and marking might need to be performed on the branch router's LAN edge in the ingress direction [13].

Fig.2.4 CBWFQ & LLQ global configuration and policy application on the Interface

```
HQ#sh run
!
class-map match-all VOICE
 match ip dscp ef
class-map match-all ROUTING
 match ip dscp cs6
class-map match-any CALL-SIGNALING
 match ip dscp cs3
 match ip dscp af31
!
policy-map BRANCH-LAN-EDGE-OUT
 class class-default
  set cos dscp
policy-map BRANCH-WAN-EDGE
 class VOICE
  priority percent 45
 class CALL-SIGNALING
  bandwidth percent 5
 class ROUTING
  bandwidth percent 3
 class class-default
  fair-queue
```

```
HQ#sh run
HQ#sh run

interface GigabitEthernet0/0.100
 description VVLAN SUBNET
 encapsulation dot1Q 100
 ip address 135.33.66.1 255.255.255.0
 service-policy output BRANCH-LAN-EDGE-OUT
!
!

interface Serial0/0/0
 ip address 172.16.22.2 255.255.255.248
 clock rate 2000000
 max-reserved-bandwidth 100
 service-policy output BRANCH-WAN-EDGE
 ip rtp priority 2000 16383 45
```

A four class QoS design:

Voice, Call-Signaling, Routing and (generic) data is configured on each aggregator router. Fig.2.4, illustrates the four class CBWFQ-LLQ QoS configuration model. Voice is identified by DSCP EF, which is set by default on Cisco IP phones. When identified, VoIP is admitted into the LLQ, which in this model is set to the maximum value of 45 percent of the link. Call admission control (CAC) correspondingly should be assigned to this link by dividing the allocated bandwidth by the voice codec including Layer 2 overhead to determine how many calls can be permitted simultaneously over this link. Because ip rtp priority command is to provide a strict priority for voice traffic, it also should be factored into the CAC calculation. Call-Signaling traffic also is marked on the IP phones to AF31 currently, but it will be migrated to CS3, per the QoS Baseline and requires a relatively small but dedicated bandwidth guarantee. Routing protocols traffic is also marked as CS6 and small dedicated bandwidth because this leaves cycles available to respond efficiently to routing updates. All other data is fair-queued within class-default.

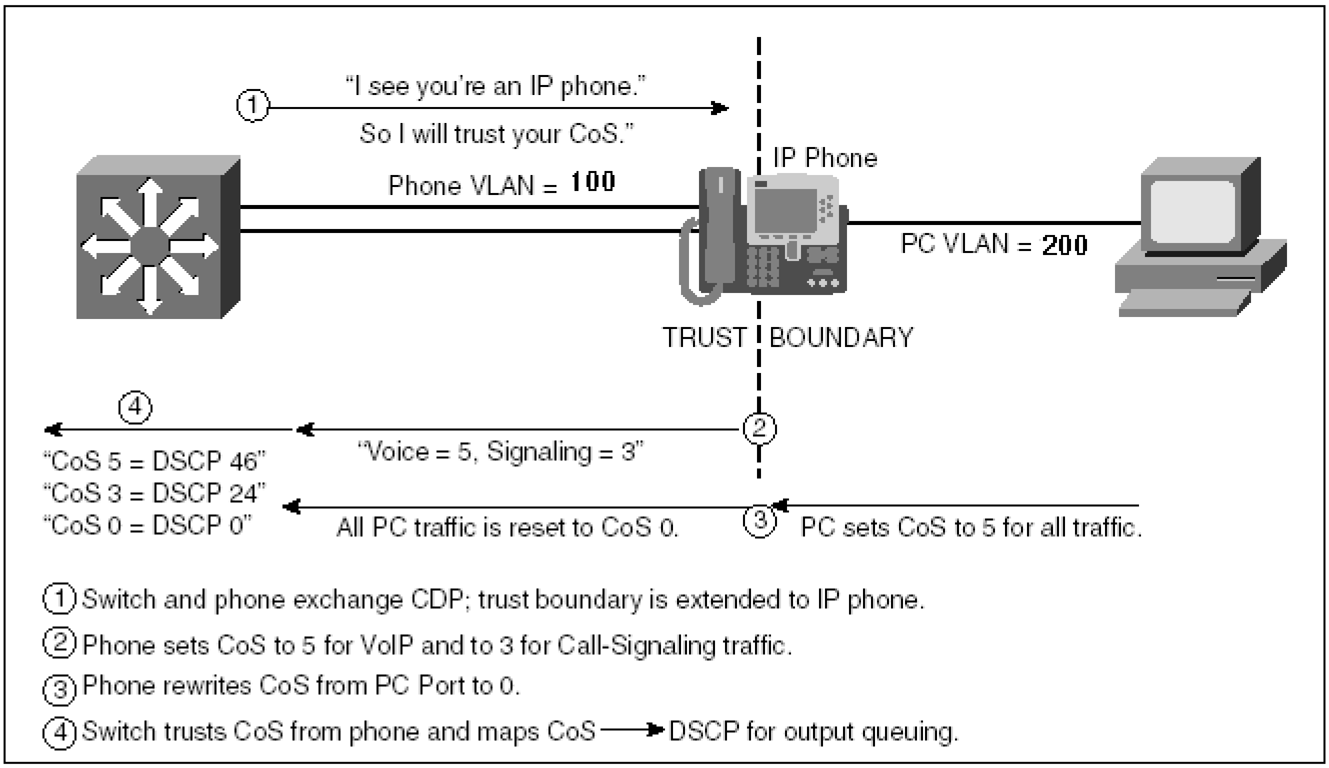
.The sum of all bandwidth allocation on an interface should not exceed 75 percent of the available bandwidth on an interface. The remaining 25 percent of bandwidth is used for overhead, including Layer 2 overhead, control traffic, and best-effort traffic [16]. If more than 75 percent is needed for CBWFQ and IP RTP Priority, the *max-reserved- bandwidth* command will be used. The percent argument specifies the maximum percentage of the total interface bandwidth that can be used by CBWFQ classes and IP RTP Priority. Fig2.4 shows the applied outbound policies and other QoS applications like ip rtp priority and Max-bandwidth on the interface.

Some applications are completely symmetrical and require identical bandwidth provisioning on both ends of the WAN link (Branch-to-WAN and HQ-to-WAN). Furthermore, having symmetrical polices on both sides of the WAN links greatly simplifies QoS policy deployment and management, which is an important aspect of large-scale designs [2]. The voice application in this work is completely symmetrical which uses same type of codec G7.29 and SCCP signaling protocol. Hence all the routers and switches have identical bandwidth and QoS policy deployment and management. All Branch and HQ routers and switches have identical configuration method [See Appendix-A].

2.5 Switch QoS Design

Cisco 3750 switches are capable of classification, marking and policing QoS configuration commands. Marking and policing are the only supported Per Hop Behavior (PHB) actions in the Cisco Catalyst 3750 Switch.

Fig.2.5, Switch Trust boundary consideration.



Modular QoS Command Line Interface (MQC)[18] is used to classify and mark the packet, and mark the incoming packets with the policy−map. The Cisco 3750 switch is configured with un-trusted PC with SoftPhone Model. The configuration performs tasks like trust the CoS values of the IP phone traffic, Mark DSCP value of the softphone application packets from the PC which is connected to the IP phone and untrust all other traffic from the PC. Fig.2.5, illustrates trust boundary consideration for Branch and HeadQuarter QoS design.

Fig.2.6 illustrates the QoS configuration on the switch and Fig.2.7 shows the logical process of marking SoftPhone traffic from an untrusted PC endpoint.

Fig.2.6 Switch QOS global configuration and interface policy application

```
HQ-switch#sh run
!
mls qos
mls qos map cos-dscp 0 8 16 24 32 46 48 56
!
class-map match-all Phone-Signaling
  match access-group name Ph-signal
class-map match-all  VOICE
  match access-group name IP-Voice

policy-map Phone-PC
  class VOICE
   set dscp ef
    police 128000 8000 exceed-action drop
  class Phone-Signaling
   set dscp cs3
    police 32000 8000 exceed-action policed-dscp-
transmit
  class class-default
   set dscp default
    police 5000000 8000 exceed-action policed-dscp-
transmit
ip access-list extended IP-Voice
 permit udp any any range 16384 32767
ip access-list extended Ph-Signal
 permit tcp any any range 2000 2002
```

```
HQ-switch#sh run
!
interface GigabitEthernet1/0/20
 switchport trunk encapsulation
dot1q
 switchport trunk native vlan 200
 switchport mode trunk
 switchport voice vlan 100
 service-policy input Phone-PC
 speed 10
 spanning-tree portfast
!
!
interface GigabitEthernet1/0/23
 switchport trunk encapsulation
dot1q
 switchport trunk native vlan 200
 switchport mode trunk
 switchport voice vlan 100
 service-policy input Phone-PC
 speed 10
 spanning-tree portfast
!
```

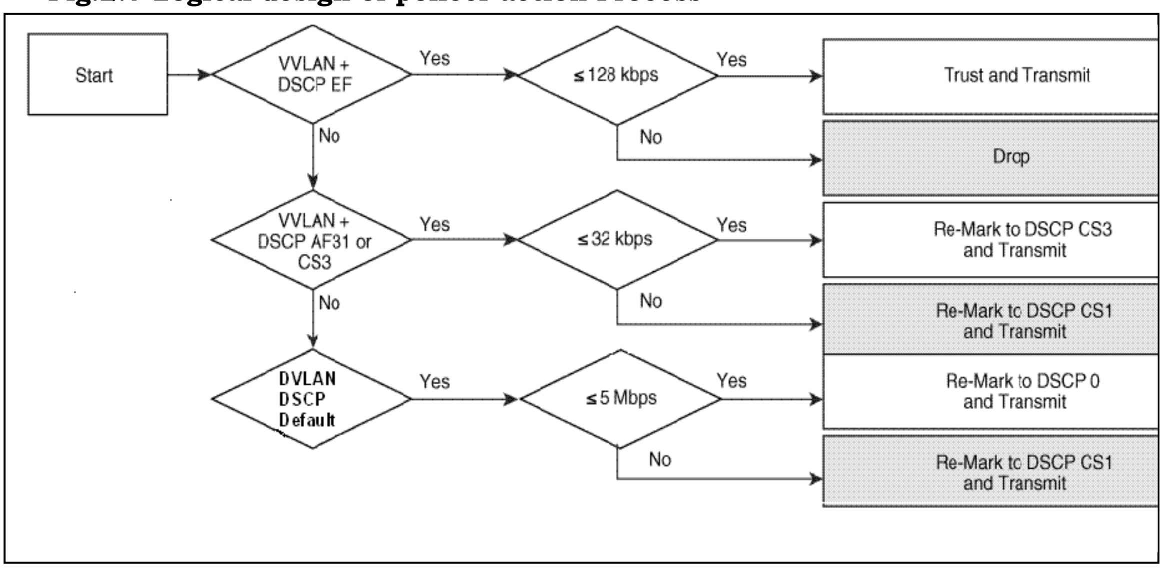
Two PHB actions are performed on the traffic matching of all classes. One is marking and the second one is policing.

Class-VOICE: The DSCP value for Class-VOICE traffic is trusted. The incoming packets that match Class-Phone-Signaling are marked with the DSCP value EF, and the CoS value is derived from the DSCP–CoS table which is 5. Then, the Class Class-VOICE traffic is policed at the rate of 128 Kbps. The excessive packets are dropped by the policer.

Class-Phone-Signaling: The incoming packets that match Class-Phone-Signaling are marked with the DSCP value CS3, and the CoS value is derived from the DSCP−CoS table which is 3. Then, the Class C traffic is policed at the rate of 32 Mbps. The excessive packets are dropped by the policer.

Class-default: The incoming packets that match class-default are marked with the DSCP value default =0 and the CoS value is derived from the DSCP−CoS table which is 0. Then, the class-default traffic is policed at the rate of 5 Mbps. The excessive packets are dropped by the policer.

Fig.2.7 Logical design of policer action Process



2.6 Measurements Design

When ever a large-scale VoIP network is managed, there must be the necessary tools to objectively monitor and report on the voice quality in the network [13]. It is not feasible to rely on user feedback alone, because it is often subjective and incomplete. Voice quality problems typically stem from network QoS problems.

Wireshark network monitoring tool is used for data capturing and to do the analysis based on the outputs. Wireshark is capable to capture Voice traffic and interpret QoS measurement parameters like RTP packet interpretation, packet loss, Jitter, and bandwidth utilization statistics.

In this work, the simulated network is deployed in MINT Lab. No long distance connectivity for remote site access. Hence, Propagation delay and delays caused by distance are not involved. To include this distance connectivity issue in the simulated network, the link bandwidth is congested for about 90-95% of the available link on interfaces. To congest all the router interfaces at the same level, equal data traffic is generated from each of Computers in the station.

Three step phone call measurements are taken: 1. only voice traffic is passing in the network, 2. after the link is congested, 3. after QoS is implemented and the link is congested.

As discussed before, the network topology and voice applications from branch to branch and branch to HQ are symmetrical. How ever for this experiment, I initiated call from each single phone to the rest of other phones located into different site (72 phone calls), then choose samples for analysis purpose. Real Time Protocol (RTP) packets are captured from phone conversations.

Real-time Transport Protocol (RTP) provides end-to-end network transport functions suitable for applications transmitting real-time data, such as audio, video or simulation data, over multicast or unicast network services. RTP does not address resource reservation and does not guarantee quality-of-service for real-time services. Typically, RTP uses UDP as its transport protocol. RTP does not have a well known UDP port the ports are allocated dynamically and then signaled using a different protocol such as SCCP, SIP or H245.

CHAPTER THREE

3.1 Data analysis and Interpretation

This chapter deals with the analysis and interpretation of experimented data. Based on measurement methodology stated in chapter 2, data was collected from live phone call setups. From HeadQuarter to Branch office, Branch offices to HeadQuarter, and Branch to Branch IP phones. The call was performed during three different link situations. While only voice traffic passing through the link, after the link is congested but no QoS, and congested link with implemented QoS.

3.2 Default VoIP quality characteristics and recommendations.

Voice quality is measured by the three major quality factors of QoS performance metrics: packet loss, delay and Jitter. Before the analysis of experiment result, the following default VOIP quality factor characteristics need to be considered.

Loss: VoIP networks typically are designed for very close to 0 percent VoIP packet loss, with the only actual packet loss being due to L2 bit errors or network failures [12,13].

Jitter: extensive lab testing has shown that voice quality degrades significantly when jitter consistently exceeds 30 ms, this 30 ms value can be used as a jitter target [12].

Bandwidth: The bandwidth that VoIP streams consume in bits per second is calculated by adding the VoIP sample payload in bytes to the 40-byte IP, UDP, and RTP headers, multiplying this value by 8 to convert it to bits, and then multiplying again by the packetization rate default of 50 packets per second. IP Bandwidth = (Bytes of Voice Payload + IP header) x 8bits per byte x IP packets generated per second of voice.

Table 3-1 Details the bandwidth per VoIP flow in both G.711 and G.729 at a default packetization rate of 50 packets per second (pps) and at a custom packetization rate of 33 pps.

Table 3-1, Bandwidth per VoIP flow of G.711 and G.729 codec.

Bandwidth Consumption	Packetization Interval	Voice Payload in Bytes	Packets Per Second	Bandwidth Per Conversation
G.711	20ms	160	50	80 kbps
G.711	30ms	240	33	74 kbps
G.729	20ms	20	50	24 kbps
G.729	30ms	30	33	19 kbps

The International Telecommunication Union Telecommunication Standardization Sector (ITU-T) G.114 recommendation suggests no more than 150 milliseconds (ms) of end-to-end delay to maintain good voice quality [8]. Any customer's definition of good might mean more or less delay, so keep in mind that 150 ms is merely a recommendation.

Serialization delay: Serialization delay refers to the finite amount of time it takes to clock a frame onto the physical media. Within the branches or campus, this time is so infinitesimal that it is completely immaterial. Over the WAN, however, lower link speeds can cause sufficient serialization delay to adversely affect real-time streams, such as Voice or Interactive-Video [12, 13].

3.2 Quality of service Performance Analysis

3.2.1 Comparison Of captured data

For VoIP QoS experiments, 9 call setups from three different network conditions are chosen. When there was only voice traffic on the network, after the links are congested, and after QoS is configured and the links are congested. The calls are from Branch-2A to Branch-1B, Branch-2B to Headquarter, and HeadQuarter to Branch-2A. An average of 4500 packets captured from each single call conversation. The reason for large number of captured packet is, to observe the voice traffic stability and feasible effect of QoS from the network. Table-3.2 shows how the network treated the voice signal in the above three different network conditions in terms of QoS factors. Factor results are taken from the summarized RTP stream conversation of each phone call as shown in fig 3.2.

Table-3.2, Result of QoS factors from the three link bandwidth utilization.

Call setup	Only Voice Traffic			Congested link			Congested Link + QoS		
	Loss	Max Delay/ms	Mean Jitt/ms	Loss	Delay /ms	Mean Jitt/ms	Loss	Delay /ms	Mean Jitt/ms
Br2A-r1B	0(0.0%)	20.47	0.04	0(0.0)	67.83	5.47	0(0.0%)	38.90	5.37
Br2B-HQ	0(0.0%)	20.74	0.03	13(0.6%)	50.48	5.24	0(0.0%)	38.81	5.11
HQ-Br2A	0(0.0%)	21.70	0.03	9(0.4%)	55.71	5.58	0(0.0%)	38.27	5.42

Only Voice Traffic in the link:

Considering the default delay budget which is 20ms for G.729 codec, serialization delay, and other budgets mentioned in table-3.1, Voice flow without competent traffic was in a very good quality. This condition will be ideal after VoIP is deployed in a real packet switched network. Approximately the default codec characteristics are observed. Consider the first row, phone call from Br2 to Br1; no loss, 0.47 (2.35%) of additional delay from the default and 0.13% of the maximum recommended jitter are observed.

Voice in non congested link VS congested link:

The congested link shows the following additional delay and average jitter from the ideal condition.

- Br2A to Br1B: 47.63ms delay and 5.43ms jitter. Uses 45.2% of 150ms delay and 18% of 30ms jitter from the maximum tolerable recommendation.
- Br2B to HQ: 39.74ms delay, 13(0.6%) packet loss, and 5.21ms jitter. Uses 33.64% of 150ms delay and 17% of 30ms jitter from the maximum tolerable recommendation. And has 0.06% additional packet loss from the recommended 0.
- HQ-Br2A: 34.01ms delay, 9(0.4%) packet loss and 5.55ms jitter. Uses 37.13% of 150ms delay and 18. % of 30ms jitters from the maximum tolerable recommendation. And has 0.04% additional packet loss from the recommended 0.

Voice traffic quality is degraded in a congested network. However, according to the recommended standard budgets, it is human ear tolerable.

Voice in congested link with QoS VS congested link without QoS:

The ration minimizes the delay and jitter occurred during the congested link.

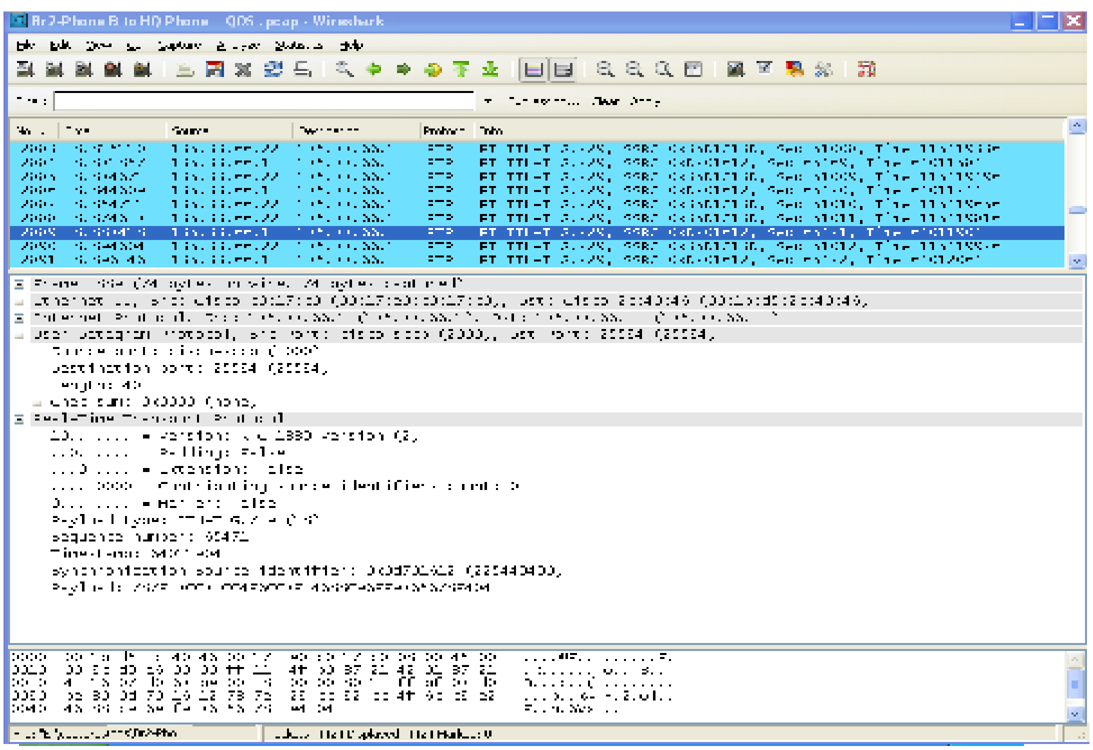
- Br2A to Br1B: 67.83 – 38.90 = 26.93ms delay and 0.10ms average jitter, delay and jitter are minimized by 39.6% and 1.8% respectively.

- Br2B to HQ: 50.48 – 38.81 = 11.67ms delay and 0.13ms average jitter, delay and jitter are minimized the by 23% and 2.48% respectively.
- HQ-Br2A: 55.71 – 38.27 = 15.44ms delay and 0.16ms average jitter. Delay and jitter are minimized the by 27.7% and 2.86% respectively.

The QoS configuration has take effect of prioritizing the voice traffic. All packet loss is recovered, and the delay difference is considerable. Regarding the average jitter, the QoS doesn't make such a big difference. Jitter is a smoothed derivative of the inter-arrival delay time [19]. So it will not get nearly as high as the delay time itself, unless fluctuations of delay time are very frequent and of high amplitude over a longer period of time.

3.2.2 Analysis of Monitored Data

Fig.3.1, Branch2-B to HQ phone conversation



The displayed packet conversation in fig.3.1 is from Branch2-B to HQ phone call setup after QoS is implemented in the VoIP network. The data was captured from the receiver phone side, HQ switch session monitor port.

The List pane of fig.3.1 shows, list of captured RTP voice packets with their source and destination phone address, the codec and sequence numbers. IP address 135.33.66.22 and SSRC 0xD701612 represent the HQ IP-Phone, and SSRC 0x35D1C13D is Branch2-B's IP-Phone. Headquarter router's LAN-Edge interface address 135.33.66.1 corresponds to the send and receive traffic of remote phone.

Synchronization source (SSRC): The source of stream of RTP packets, identified by a 32-bit numeric SSRC identifier carried in the RTP header so as not to be dependent upon the network address [20].

The Skinny Call Control Protocol (SCCP) shown in the protocol pane, sets up, tears down, or modifies connections in which RTP can transfer the audio data. Wireshark can see the same traffic and understand that it's SCCP RTP/UDP traffic. This makes it a bit more powerful in seeing what the traffic is being used for.

Fig.3.2, Summery of forward and reverse direction RTP streams

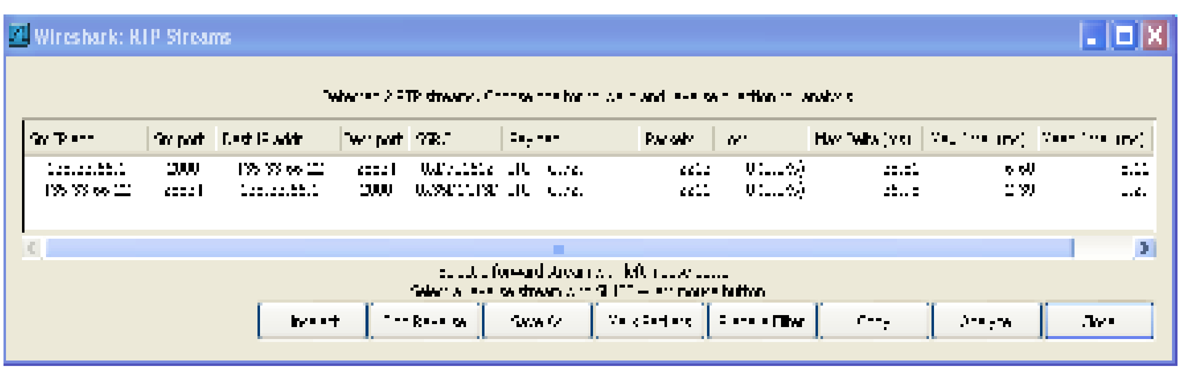
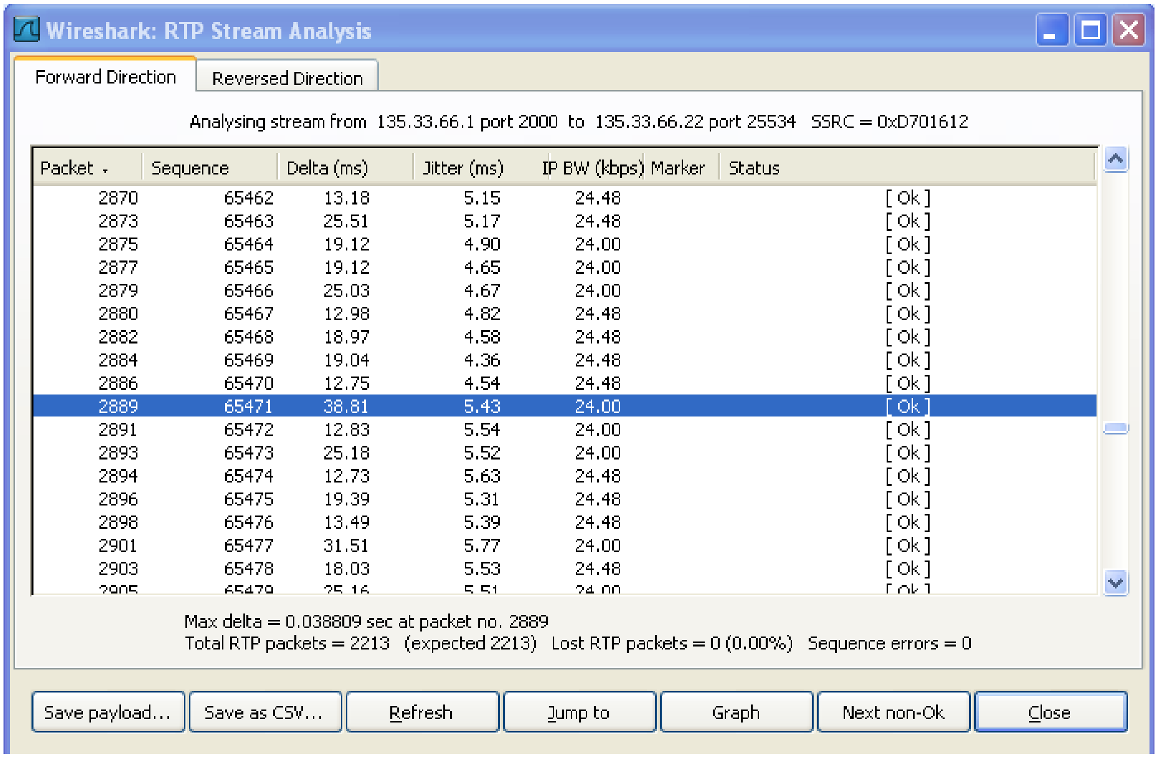


Fig.3.2 shows the summery of forward and reverse direction RTP streams analysis. The first stream has no packet loss, 38.81ms maximum delay time, 6.60ms maximum jitter time and 5.11ms mean value of jitter time as discussed in table 3.2.

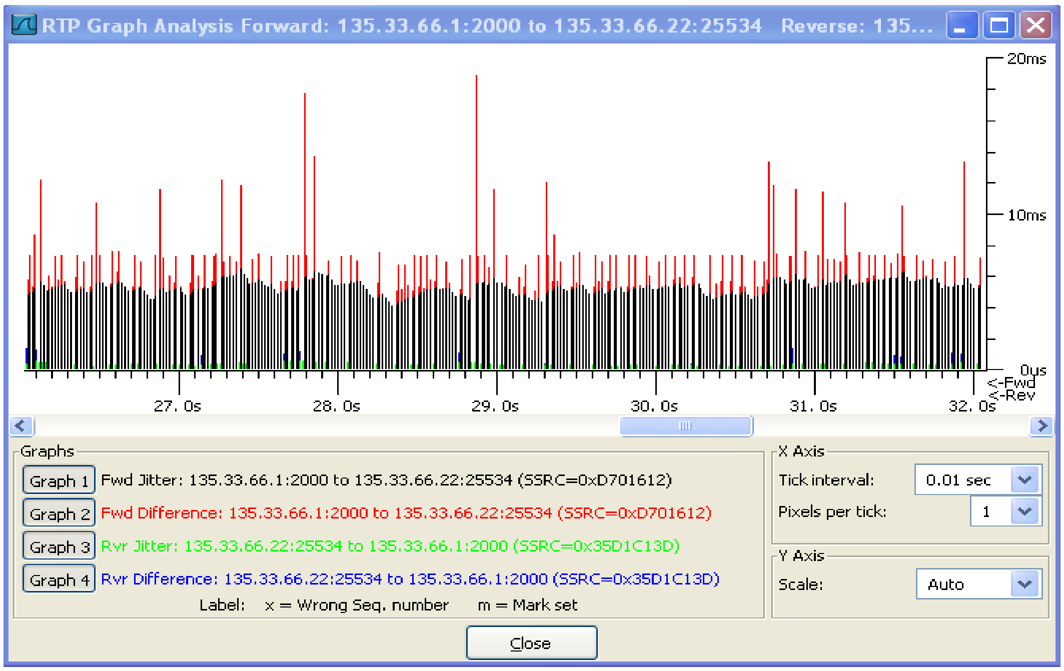
Fig.3.3, Detail analysis of forward and reverse directions RTP stream.



Packet	Sequence	Delta (ms)	Jitter (ms)	IP BW (kbps)	Marker	Status
2870	65462	13.18	5.15	24.48		[Ok]
2873	65463	25.51	5.17	24.48		[Ok]
2875	65464	19.12	4.90	24.00		[Ok]
2877	65465	19.12	4.65	24.00		[Ok]
2879	65466	25.03	4.67	24.00		[Ok]
2880	65467	12.98	4.82	24.48		[Ok]
2882	65468	18.97	4.58	24.48		[Ok]
2884	65469	19.04	4.36	24.48		[Ok]
2886	65470	12.75	4.54	24.48		[Ok]
2889	65471	38.81	5.43	24.00		[Ok]
2891	65472	12.83	5.54	24.00		[Ok]
2893	65473	25.18	5.52	24.00		[Ok]
2894	65474	12.73	5.63	24.48		[Ok]
2896	65475	19.39	5.31	24.48		[Ok]
2898	65476	13.49	5.39	24.48		[Ok]
2901	65477	31.51	5.77	24.00		[Ok]
2903	65478	18.03	5.53	24.48		[Ok]

The analysis window in fig.3.3 proposes information about the terminals engaged in the conversation, displays helpful statistics about the flow, such as the current bandwidth, latency and jitter. The highlighted line shows packet number 2889 where the highest delay time 38.81ms is registered. It also registers 5.43ms of jitter which is higher than the mean jitter value 5.11ms of the entire conversations. Bandwidth utilization is equal to the default budget that codec G.729 uses.

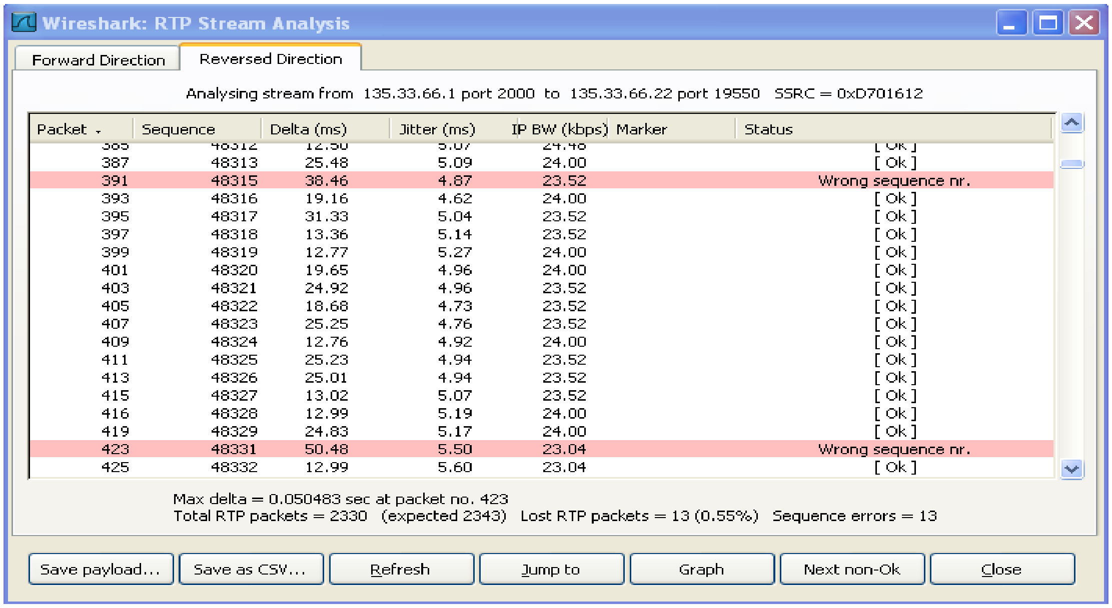
Fig.3.4, RTP analysis graph.



The RTP graph shows the forwarded and reversed direction jitter and difference between packets over the time.

Fig.3.5 is RTP stream analysis from the phone setup between Branch2 to Headquarter, before QoS is implemented. As detailed, there are 13(0.55%) packet losses due to highly congested link. The maximum delay time is from packet number 423 where the third packet loss has occurred. All odd packet numbers in this figure belong to the displayed reverse direction traffic flow. Packet number 389 and 421 are missed from the figure. Lost packets show up as having the wrong sequence number in the list pane of this capture. This loss will not be seen in the reverse direction, because the capture was taken close to the source of the conversation. The maximum delay and jitter of this RTP flow are acceptable compare to the limits 150ms delay and 30ms jitter. Even though the loss seems intolerable from the advised loss which is 0, during this experiment, it was not that hard to hear each other but the line was noisy.

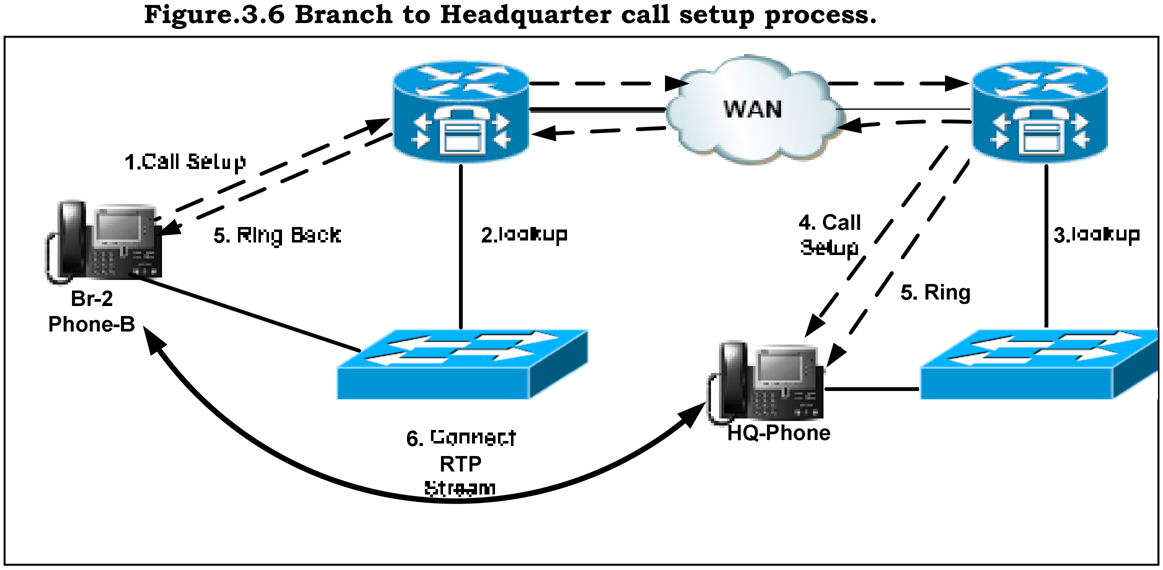
Fig.3.5, RTP Streams with packet loss.



3.3 QoS performance on Routers and switches

The call setup scenario from Branch-2 to Head Quarter passes the stapes illustrated in figure 3. Hence, all the discussed QoS factors packet loss, latency, jitter and bandwidth behavior is result of routers and switches deployment design and configuration role.

Figure.3.6 Branch to Headquarter call setup process.



The show interface and show policy interface commands are taken right after the captured phone conversation.

Fig.3.7, Headquarter and Branch-2 routers show interface Verification command

```
HQ#sh int s0/0/0
Serial0/0/0 is up, line protocol is up
  Hardware is GT96K Serial
  Internet address is 172.16.22.2/29
  MTU 1500 bytes, BW 1544 Kbit, DLY 20000 usec,
     .
  Input queue: 0/75/0/0 (size/max/drops/flushes); Total output drops: 181461
  Queueing strategy: Class-based queueing
  Output queue: 55/1000/64/181461 (size/max total/threshold/drops)
     Conversations  4/15/256 (active/max active/max total)
     Reserved Conversations 2/2 (allocated/max allocated)
     Available Bandwidth 682 kilobits/sec
  5 minute input rate 1962000 bits/sec, 306 packets/sec
  5 minute output rate 1961000 bits/sec, 307 packets/sec
     70723598 packets input, 2951104025 bytes, 0 no buffer
     Received 140867 broadcasts, 0 runts, 0 giants, 0 throttles
     0 input errors, 0 CRC, 0 frame, 0 overrun, 0 ignored, 0 abort
     67397042 packets output, 2066068052 bytes, 0 underruns
     DCD=up  DSR=up  DTR=up  RTS=up  CTS=up

BR-2#sh int s0/0/0
Serial0/0/0 is up, line protocol is up
  Hardware is GT96K Serial
  Internet address is 172.16.22.10/29
  MTU 1500 bytes, BW 1544 Kbit, DLY 20000 usec,

  Input queue: 0/75/0/0 (size/max/drops/flushes); Total output drops: 72391
  Queueing strategy: Class-based queueing
  Output queue: 49/1000/64/72391 (size/max total/threshold/drops)
     Conversations  1/11/256 (active/max active/max total)
     Reserved Conversations 2/2 (allocated/max allocated)
     Available Bandwidth 682 kilobits/sec
  5 minute input rate 1958000 bits/sec, 308 packets/sec
  5 minute output rate 1960000 bits/sec, 306 packets/sec
     26760586 packets input, 629042400 bytes, 0 no buffer
     Received 50516 broadcasts, 0 runts, 0 giants, 0 throttles
     0 input errors, 0 CRC, 0 frame, 0 overrun, 0 ignored, 0 abort
     29337580 packets output, 1080726007 bytes, 0 underruns
          DCD=up  DSR=up  DTR=up  RTS=up  CTS=up
```

The show interface verification command displays the queuing strategy, threshold statistics, bandwidth utilization and packet transfer rates. Additionally, the command indicates whether drops are occurring on the interface. As detailed in fig.3.7, 181461(0.27%) packets are dropped and 67397042 packets transferred from the total of 67578503 output packets that HQ WAN-Edge interface received, and 72391(0.25%) packets are dropped and 29337580 packets transferred from the total of 29409971 output packets that Branch-2 router WAN-Edge interface received. Both the links are utilized for about 98% and have approximately the same packets/sec transfer rate.

The show policy interface command displays a wide array of dynamic statistics, including the number of matches on a class map as a whole, the number of matches against each discrete match statement within a class map, and the number of queued or dropped packets. Fig.3.8, shows the result of HQ and Branch-2 routers show policy interface command.

Fig.3.8, HQ and Branch-2 routers show policy interface command.

```
HQ#sh policy-map int s0/0/0
 Serial0/0/0

  Service-policy output: BRANCH-WAN-EDGE

    Class-map: VOICE (match-all)
      124386 packets, 7960704 bytes
      5 minute offered rate 7000 bps, drop rate 0 bps
      Match: ip dscp ef (46)
      Queueing
        Strict Priority
        Output Queue: Conversation 264
        Bandwidth 45 (%)
        Bandwidth 694 (kbps) Burst 17350 (Bytes)
        (pkts matched/bytes matched) 7138/456832
        (total drops/bytes drops) 0/0

    Class-map: CALL-SIGNALING (match-any)
      684 packets, 84141 bytes
      5 minute offered rate 0 bps, drop rate 0 bps
        Match: ip dscp cs3 (24)
        0 packets, 0 bytes
        5 minute rate 0 bps
        Match: ip dscp af31 (26)
        684 packets, 84141 bytes
        5 minute rate 0 bps
      Queueing
        Output Queue: Conversation 265
        Bandwidth 5 (%)
        Bandwidth 77 (kbps)Max Threshold 64 (packets)
        (pkts matched/bytes matched) 684/84141
        (depth/total drops/no-buffer drops) 0/0/0
```

```
Class-map: ROUTING (match-all)
      58528 packets, 4889504 bytes
      5 minute offered rate 0 bps, drop rate 0 bps
      Match: ip dscp cs6 (48)
      Queueing
        Output Queue: Conversation 266
        Bandwidth 3 (%)
        Bandwidth 46 (kbps)Max Threshold 64 (packets)
        (pkts matched/bytes matched) 5202/421236
        (depth/total drops/no-buffer drops) 0/0/0

    Class-map: class-default (match-any)
      12358138 packets, 7299497751 bytes
      5 minute offered rate 1954000 bps, drop rate 7000
bps
      Match: any
      Queueing
        Flow Based Fair Queueing
        Maximum Number of Hashed Queues 256
        (total queued/total drops/no-buffer drops)
57/24798/0

HQ#sh policy-map int gi0/0.100
 GigabitEthernet0/0.100

  Service-policy output: BRANCH-LAN-EDGE-OUT
    Class-map: class-default (match-any)
      257408 packets, 21101182 bytes
      5 minute offered rate 11000 bps, drop rate 0 bps
      Match: any
      QoS Set
        cos dscp
          Packets marked 257408
```

```
BR_2#sh policy-map int s0/0/0
 Serial0/0/0

  Service-policy output: BRANCH-WAN-EDGE

    Class-map: VOICE (match-all)
      96968 packets, 6205952 bytes
      5 minute offered rate 17000 bps, drop rate 0 bps
      Match: ip dscp ef (46)
      Queueing
        Strict Priority
        Output Queue: Conversation 264
        Bandwidth 45 (%)
        Bandwidth 694 (kbps) Burst 17350 (Bytes)
        (pkts matched/bytes matched) 74303/4755392
        (total drops/bytes drops) 0/0

    Class-map: CALL-SIGNALING (match-any)
      492 packets, 63027 bytes
      5 minute offered rate 0 bps, drop rate 0 bps
      Match: ip dscp cs3 (24)
        0 packets, 0 bytes
        5 minute rate 0 bps
      Match: ip dscp af31 (26)
        492 packets, 63027 bytes
        5 minute rate 0 bps
      Queueing
        Output Queue: Conversation 265
        Bandwidth 5 (%)
        Bandwidth 77 (kbps)Max Threshold 64 (packets)
        (pkts matched/bytes matched) 492/63027
        (depth/total drops/no-buffer drops) 0/0/0
```

```
    Class-map: ROUTING (match-all)
      55327 packets, 4633592 bytes
      5 minute offered rate 0 bps, drop rate 0 bps
      Match: ip dscp cs6 (48)
      Queueing
        Output Queue: Conversation 266
        Bandwidth 3 (%)
        Bandwidth 46 (kbps)Max Threshold 64 (packets)
        (pkts matched/bytes matched) 2108/173596
        (depth/total drops/no-buffer drops) 0/0/0

    Class-map: class-default (match-any)
      15065977 packets, 8525366296 bytes
      5 minute offered rate 1918000 bps, drop rate
4000 bps
      Match: any
      Queueing
        Flow Based Fair Queueing
        Maximum Number of Hashed Queues 256
        (total queued/total drops/no-buffer drops)
26/27044/0
BO_2#sh policy-map int gi0/0.100
 GigabitEthernet0/0.100

  Service-policy output: BRANCH-LAN-EDGE-OUT

    Class-map: class-default (match-any)
      317974 packets, 26042136 bytes
      5 minute offered rate 19000 bps, drop rate 0 bps
      Match: any
      QoS Set
        cos dscp
```

In fig3.8, the Voice class map and Call-Signaling class map are receiving matches on their classification criteria DSCP EF and DSCP CS3/AF31, respectively. However, because Cisco IP Telephony products currently mark Call-Signaling traffic to DSCP AF31, Call-Signaling traffic is matching only on DSCP AF31 in the above figure. Strict Priority, the ip-rtp priority configuration result is included under the Voice class map.

The last line of every class map output indicates whether any drops are occurring on traffic classes. In this figure, there are no drops in the Voice or Call-Signaling, and Routing classes, which is the desired behavior. Drops are occurred in class-default but this is expected when the interface is congested which is the trigger to engage queuing. The total drops statistics indicates adequate bandwidth has been assigned to the mission critical traffic classes. Figure3.9, details the class default traffic drops at the WAN-Edge of HQ router before they entered to the WAN.

Fig.3.9, Policy action on non prioritized data traffic.

```
HQ#sh queue s0/0/0
  Input queue: 0/75/0/0 (size/max/drops/flushes); Total output drops: 121088
  Queueing strategy: Class-based queueing
  Output queue: 61/1000/64/121088 (size/max total/threshold/drops)
     Conversations  4/11/256 (active/max active/max total)
     Reserved Conversations 2/2 (allocated/max allocated)
     Available Bandwidth 682 kilobits/sec

  (depth/weight/total drops/no-buffer drops/interleaves) 29/32384/1895/0/0
  Conversation 227, linktype: ip, length: 1504
  source: 135.33.166.22, destination: 135.33.165.22, id: 0x492D, ttl: 63,
  TOS: 0 prot: 6, source port 46411, destination port 22

  (depth/weight/total drops/no-buffer drops/interleaves) 2/32384/1112/0/0
  Conversation 249, linktype: ip, length: 1504
  source: 135.33.166.21, destination: 135.33.167.22, id: 0x1F08, ttl: 63,
  TOS: 0 prot: 6, source port 60712, destination port 22

  (depth/weight/total drops/no-buffer drops/interleaves) 24/32384/2153/0/0
  Conversation 129, linktype: ip, length: 1504
  source: 135.33.166.21, destination: 135.33.165.21, id: 0x9CEF, ttl: 63,
  TOS: 0 prot: 6, source port 40705, destination port 22

  (depth/weight/total drops/no-buffer drops/interleaves) 6/32384/1135/0/0
  Conversation 163, linktype: ip, length: 1504
  source: 135.33.166.22, destination: 135.33.167.22, id: 0x670B, ttl: 63,
  TOS: 0 prot: 6, source port 49151, destination port 22
```

Fig.3.10, Marking, mapping and policy action process on the switch port.

```
BR2-switch#sh mls qos interface policers
GigabitEthernet1/0/20
policymap=Phone-PC
type=Single, id=1 rate=128000, qlimit=8000, drop=0
 markdown dscp=0 type=Single, id=2 rate=32000, qlimit=8000, drop=0
 markdown dscp=0 type=Single, id=3 rate=5000000, qlimit=8000, drop=0
 markdown dscp=0
```

The show policy interface command on the Catalyst 3750 shows zeros for the counters associated with class-map match criteria. Referring the Cisco [12, 18], this has been reported as a bug. In other words, all counters currently are frozen at zero. Fig.3.10 shows the applied policy-map and its marking, mapping and policy action process on the switch port connected to IP-Phones.

CHAPTER FOUR

CONCLUSION

Providing reliable, high-quality voice communication over a network designed for data communications is a complex engineering challenge. To provide a high quality voice over IP, the choice of codec, call signaling protocol and QoS technique are important factors. In this paper, Class Based Waited Fair Queue (CBWFQ) Quality of Service mechanism has been implemented and experimented. This technique guaranteed the voice flow from loss; shorten the delay time or low latency, and shows slight difference with the jitter. In favor of Voice, two Per-Hop-Behaviors (PHB) applied in this technique: Expedited forwarding (EF) and Assured Forwarding (AF) provide low loss, low latency service and assured the allocated bandwidth service respectively. In addition to its powerful decision, the allocated bandwidth percent could be utilized by other applications if there is sufficient bandwidth for voice or if there is no voice traffic on the network. However, the voice performance success of this technique is; on the expense of routers' overhead during traffic classification, marking and scheduling process of CBWFQ technique, and on the expense of the data traffic loss. The experiment was for an average of 45seconds phone calls plus there is no long distance connection. But in the real world phone calls, we usually stay online for minutes. If there is similar link congestion with long distance interconnected routers, delay will be increasing, the routers performance will be degraded, as well as high volume of data packet loss will occur. All these reasons and other VoIP quality issues like the QoS design and implementation are make the idea of implementing QoS on VoIP network an open area of research.

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APPENDEX-A

Routers and Switches Running Configuration

1. Branch-1

Router Configuration

```
BR-1#sh run
Building configuration...

Current configuration : 3903 bytes
!
! Last configuration change at 14:49:36 mountai Tue Jun 17 2008
! NVRAM config last updated at 14:49:37 mountai Tue Jun 17 2008
!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname BR-1
!
boot-start-marker
boot-end-marker
!
enable secret 5 $1$6Ela$4CWyPUEiPy9SGL.3nSdCF.
!
no aaa new-model
!
resource policy
!
clock timezone mountia -7
clock summer-time mountain recurring
!
!
ip cef
no ip dhcp use vrf connected
ip dhcp excluded-address 135.33.67.1 135.33.67.20
ip dhcp excluded-address 135.33.167.1 135.33.167.20
!
ip dhcp pool BR1-Data
   network 135.33.167.0 255.255.255.0
   default-router 135.33.167.1
   option 150 ip 135.33.167.1
!
ip dhcp pool BR1-Voice
   network 135.33.67.0 255.255.255.0
   default-router 135.33.67.1
   option 150 ip 135.33.67.1
!
!
no ip domain lookup
```

```
!
!
voice-card 0
 no dspfarm
!
class-map match-all VOICE
 match ip dscp ef
class-map match-all ROUTING
 match ip dscp cs6
class-map match-any CALL-SIGNALING
 match ip dscp cs3
 match ip dscp af31
!
!
policy-map BRANCH-LAN-EDGE-OUT
 class class-default
  set cos dscp
policy-map BRANCH-WAN-EDGE
 class VOICE
  priority percent 45
 class CALL-SIGNALING
  bandwidth percent 5
 class ROUTING
  bandwidth percent 3
 class class-default
  fair-queue
!
interface Loopback0
 ip address 1.1.1.3 255.255.255.255
!
interface GigabitEthernet0/0
 no ip address
 duplex auto
 speed 10
!
interface GigabitEthernet0/0.1
 encapsulation dot1Q 200 native
 ip address 135.33.167.1 255.255.255.0
!
interface GigabitEthernet0/0.100
 description VVLAN SUBNET
 encapsulation dot1Q 100
 ip address 135.33.67.1 255.255.255.0
 service-policy output BRANCH-LAN-EDGE-OUT
!
interface GigabitEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed 10
!
interface FastEthernet0/1/0
!
interface FastEthernet0/1/1
!
interface FastEthernet0/1/2
!
```

```
interface FastEthernet0/1/3
!
interface Serial0/0/0
 ip address 172.16.22.10 255.255.255.248
 clock rate 2000000
 max-reserved-bandwidth 100
 service-policy output BRANCH-WAN-EDGE
 ip rtp priority 2000 16383 45
!
interface Vlan1
 no ip address
!
router ospf 1
 router-id 1.1.1.3
 log-adjacency-changes
 network 1.1.1.3 0.0.0.0 area 0
 network 135.33.67.0 0.0.0.255 area 0
 network 135.33.167.0 0.0.0.255 area 0
 network 172.16.22.8 0.0.0.7 area 0
!
!
ip http server
no ip http secure-server
!
snmp-server community private RW
snmp-server community public RW
!
!
tftp-server flash:TERM41.7-0-3-0S.loads
!
control-plane
!
dial-peer voice 1 pots
 destination-pattern 1001
!
dial-peer voice 100 voip
 preference 1
 destination-pattern 3002
 session target ipv4:1.1.1.2
!
dial-peer voice 1000 voip
 preference 1
 destination-pattern 2001
 session target ipv4:1.1.1.4
!
dial-peer voice 2 pots
 destination-pattern 1002
!
dial-peer voice 1001 voip
 preference 1
 destination-pattern 3002
 session target ipv4:1.1.1.2
!
dial-peer voice 101 voip
 preference 1
 destination-pattern 2002
 session target ipv4:1.1.1.4
```

```
!
dial-peer voice 1100 voip
 preference 1
 destination-pattern 2002
 session target ipv4:1.1.1.4
!
dial-peer voice 111 voip
 preference 1
 destination-pattern 2001
 session target ipv4:1.1.1.4
!
telephony-service
 no auto-reg-ephone
 load 7941 TERM41.7-0-3-0S.loads
 max-ephones 6
 max-dn 10
 ip source-address 135.33.67.1 port 2000
 time-zone 7
 max-conferences 8 gain -6
 transfer-system full-consult
 create cnf-files version-stamp 7960 Jun 17 2008 14:48:47
!
ephone-dn  1
 number 1001
 name A
!
!
ephone-dn  2
 number 1002
 name B
!
!
ephone  1
 mac-address 001B.D512.F3C3
 type 7941
 button  1:1
!
ephone  2
 mac-address 001B.D512.6D5E
 type 7941
 button  1:2

line con 0
line aux 0
line vty 0 4
 login
!
scheduler allocate 20000 1000
ntp master 10
ntp update-calendar
ntp peer 1.1.1.1
ntp peer 1.1.1.2
ntp peer 1.1.1.4
!
end
```

Switch Configuration

```
BR1-Switch#sh run
Building configuration...

Current configuration : 3296 bytes
!
! Last configuration change at 01:11:45 UTC Sun Jun 22 2008
! NVRAM config last updated at 01:05:32 UTC Sun Jun 22 2008
!
version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname Br1-Switch
!
enable secret 5 $1$9tiE$TUmzNqBVk9ee9lS1NacUD/
!
no aaa new-model
switch 1 provision ws-c3750g-24ps
ip subnet-zero
!
!
mls qos map cos-dscp 0 8 16 24 32 46 48 56
mls qos
!
!
no file verify auto
spanning-tree mode pvst
spanning-tree extend system-id
!
vlan internal allocation policy ascending
!
class-map match-all Phone-Signaling
  match access-group name Ph-signal
class-map match-all VOICE
  match access-group name IP-VOICE
!
!
policy-map Phone-PC
  class VOICE
   set dscp ef
    police 128000 8000 exceed-action policed-dscp-transmit
  class Phone-Signaling
   set dscp cs3
    police 32000 8000 exceed-action policed-dscp-transmit
  class class-default
   set dscp default
    police 5000000 8000 exceed-action policed-dscp-transmit
!
!
!
interface GigabitEthernet1/0/1
!
```

```
interface GigabitEthernet1/0/17
 description Monitor port
 switchport mode dynamic desirable
!
interface GigabitEthernet1/0/19
!
interface GigabitEthernet1/0/20
 switchport trunk encapsulation dot1q
 switchport trunk native vlan 200
 switchport mode trunk
 switchport voice vlan 100
 service-policy input Phone-PC
 speed 10
 spanning-tree portfast
!
interface GigabitEthernet1/0/21
 !
interface GigabitEthernet1/0/22
 !
interface GigabitEthernet1/0/23
 switchport trunk encapsulation dot1q
 switchport trunk native vlan 200
 switchport mode trunk
 switchport voice vlan 100
 service-policy input Phone-PC
 speed 10
 spanning-tree portfast
!
interface GigabitEthernet1/0/24
 switchport trunk encapsulation dot1q
 switchport trunk native vlan 200
 switchport mode trunk
 switchport voice vlan 100
 speed 10
 spanning-tree portfast
!
interface GigabitEthernet1/0/25
!
interface GigabitEthernet1/0/26
!
interface GigabitEthernet1/0/27
!
interface GigabitEthernet1/0/28
!
interface Vlan1
 no ip address
!
interface Vlan100
 ip address 135.33.67.2 255.255.255.0
 no ip route-cache
 no ip mroute-cache
!
interface Vlan200
 ip address 135.33.167.2 255.255.255.0
 no ip route-cache
 no ip mroute-cache
!
```

```
ip classless
ip http server
!
!
ip access-list extended IP-VOICE
 permit udp any any range 16384 32767
ip access-list extended Ph-Signal
 permit tcp any any range 2000 2002
!
!
control-plane
!
!
line con 0
line vty 0 4
 no login
 length 0
line vty 5 15
 no login
!
!
monitor session 1 source interface Gi1/0/20 - 21 , Gi1/0/23
monitor session 1 destination interface Gi1/0/17
ntp clock-period 36028702
ntp server 1.1.1.3
end
=======================================================
```

2. Branch-2

Router Configuration

```
Br-2#sh run
Building configuration...

Current configuration : 3653 bytes
!
! Last configuration change at 14:49:10 mountai Tue Jun 17 2008
! NVRAM config last updated at 14:49:12 mountai Tue Jun 17 2008
!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname BO_2
!
boot-start-marker
boot-end-marker
!
enable secret 5 $1$xo/2$Qmdx/DSAhjheKH/lLA/L11
!
no aaa new-model
!
resource policy
!
memory-size iomem 10
```

```
clock timezone mountia -7
clock summer-time mountain recurring
!
!
ip cef
no ip dhcp use vrf connected
ip dhcp excluded-address 135.33.65.1 135.33.65.20
ip dhcp excluded-address 135.33.165.1 135.33.165.20
!
ip dhcp pool HQ-Voice
   network 135.33.65.0 255.255.255.0
   default-router 135.33.65.1
   option 150 ip 135.33.65.1
!
ip dhcp pool HQ-Data
   network 135.33.165.0 255.255.255.0
   default-router 135.33.165.1
   option 150 ip 135.33.165.1
!
!
ip multicast-routing
!
!
voice-card 0
 dspfarm
!
class-map match-all VOICE
 match ip dscp ef
class-map match-all ROUTING
 match ip dscp cs6
class-map match-any CALL-SIGNALING
 match ip dscp cs3
 match ip dscp af31
!
!
policy-map BRANCH-LAN-EDGE-OUT
 class class-default
  set cos dscp
policy-map BRANCH-WAN-EDGE
 class VOICE
  priority percent 45
 class CALL-SIGNALING
  bandwidth percent 5
 class ROUTING
  bandwidth percent 3
 class class-default
  fair-queue
!
interface Loopback0
 ip address 1.1.1.4 255.255.255.255
!
interface GigabitEthernet0/0
 no ip address
 ip pim dense-mode
 duplex auto
 speed 10
!
```

```
interface GigabitEthernet0/0.1
 description DVLAN SUBNET
 encapsulation dot1Q 200 native
 ip address 135.33.165.1 255.255.255.0
!
interface GigabitEthernet0/0.100
 description VVLAN SUBNET
 encapsulation dot1Q 100
 ip address 135.33.65.1 255.255.255.0
 service-policy output BRANCH-LAN-EDGE-OUT
!
interface GigabitEthernet0/1
  duplex auto
 speed 10
!
interface FastEthernet0/1/0
!
interface FastEthernet0/1/1
!
interface FastEthernet0/1/2
!
interface FastEthernet0/1/3
!
interface Serial0/0/0
 ip address 172.16.22.18 255.255.255.248
 clock rate 2000000
 max-reserved-bandwidth 100
 service-policy output BRANCH-WAN-EDGE
 ip rtp priority 2000 16383 45
!
interface Vlan1
 no ip address
!
router ospf 1
 log-adjacency-changes
 network 1.1.1.4 0.0.0.0 area 0
 network 135.33.65.0 0.0.0.255 area 0
 network 135.33.165.0 0.0.0.255 area 0
 network 172.16.22.16 0.0.0.7 area 0
!
!
ip http server
no ip http secure-server
!
!
!
tftp-server flash:TERM41.7-0-3-0S.loads
!
control-plane
!
dial-peer voice 1 pots
 destination-pattern 1001
!
dial-peer voice 888 voip
 preference 1
 destination-pattern 3002
 session target ipv4:1.1.1.2
```

```
!
dial-peer voice 88 voip
 preference 1
 destination-pattern 1001
 session target ipv4:1.1.1.3
!
dial-peer voice 2 pots
 destination-pattern 1002
!
dial-peer voice 8888 voip
 preference 1
 destination-pattern 3002
 session target ipv4:1.1.1.2
!
dial-peer voice 880 voip
 preference 1
 destination-pattern 1002
 session target ipv4:1.1.1.3
!
!
!
telephony-service
 no auto-reg-ephone
 load 7941 TERM41.7-0-3-0S.loads
 max-ephones 6
 max-dn 10
 ip source-address 135.33.65.1 port 2000
 max-conferences 8 gain -6
 transfer-system full-consult
 create cnf-files version-stamp 7960 Jun 17 2008 14:49:08
!
!
ephone-dn  1
 number 2001
 name A
!
!
ephone-dn  2
 number 2002
 name B
!
!
ephone  1
 mac-address 001B.D512.3D35
 type 7941
 button  1:1
!
!
!
ephone  2
 mac-address 001B.D512.4E3A
 type 7941
 button  1:2
!
!
!
line con 0
```

```
line aux 0
line vty 0 4
 login
!
scheduler allocate 20000 1000
ntp clock-period 17179906
ntp server 1.1.1.3
!
end
```

Switch Configuration

```
BR2-switch#sh run
Building configuration...

Current configuration : 3575 bytes
!
! Last configuration change at 19:11:17 mountai Sat Jun 21 2008
! NVRAM config last updated at 15:29:44 mountai Tue Jun 17 2008
!
version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname BR2-switch
!
enable secret 5 $1$7F0y$1zROS6cqOlydJhrA6vVcJ1
enable password mine
!
no aaa new-model
clock timezone mountia -7
clock summer-time mountain recurring
switch 1 provision ws-c3750g-24ps
ip subnet-zero
!
no ip igmp snooping
!
mls qos map cos-dscp 0 8 16 24 32 46 48 56
mls qos
!
!
no file verify auto
spanning-tree mode pvst
spanning-tree extend system-id
!
vlan internal allocation policy ascending
!
class-map match-all Phone-Signaling
  match access-group name Ph-signal
class-map match-all VOICE
  match access-group name IP-VOICE
!
!
```

```
policy-map Phone-PC
  class VOICE
   set dscp ef
    police 128000 8000 exceed-action policed-dscp-transmit
  class Phone-Signaling
   set dscp cs3
    police 32000 8000 exceed-action policed-dscp-transmit
  class class-default
   set dscp default
    police 5000000 8000 exceed-action policed-dscp-transmit
!
!
!
interface GigabitEthernet1/0/1
 spanning-tree portfast
!
interface GigabitEthernet1/0/17
 description Monitor port
 switchport mode dynamic desirable
!
interface GigabitEthernet1/0/18
!
interface GigabitEthernet1/0/19
!
interface GigabitEthernet1/0/20
 switchport trunk encapsulation dot1q
 switchport trunk native vlan 200
 switchport mode trunk
 switchport voice vlan 100
 service-policy input Phone-PC
 speed 10
 spanning-tree portfast
!
interface GigabitEthernet1/0/21

!
interface GigabitEthernet1/0/22

!
interface GigabitEthernet1/0/23
 switchport trunk encapsulation dot1q
 switchport trunk native vlan 200
 switchport mode trunk
 switchport voice vlan 100
 service-policy input Phone-PC
 speed 10
 spanning-tree portfast
!
interface GigabitEthernet1/0/24
 switchport trunk encapsulation dot1q
 switchport trunk native vlan 200
 switchport mode trunk
 switchport voice vlan 100
 speed 10
 spanning-tree portfast
!
interface GigabitEthernet1/0/25
```

```
!
interface GigabitEthernet1/0/26
!
interface GigabitEthernet1/0/27
!
interface GigabitEthernet1/0/28
!
interface Vlan1
 no ip address
!
interface Vlan20
 no ip address
!
interface Vlan100
 ip address 135.33.65.2 255.255.255.0
 no ip route-cache
 no ip mroute-cache
!
interface Vlan200
 ip address 135.33.165.2 255.255.255.0
 no ip route-cache
 no ip mroute-cache
!
ip classless
ip http server
!
!
ip access-list extended IP-VOICE
 permit udp any any range 16384 32767
ip access-list extended Ph-Signal
 permit tcp any any range 2000 2002
!
!
control-plane
!
!
line con 0
line vty 0 4
 no login
 length 0
line vty 5 15
 no login
!
!
monitor session 1 source interface Gi1/0/20 - 21 , Gi1/0/23
monitor session 1 destination interface Gi1/0/17
ntp clock-period 36028702
ntp server 1.1.1.3
end
```
===

3.HeadQuarter

Router configuration

```
HQ#sh run
Building configuration...

Current configuration : 3681 bytes
!
! Last configuration change at 14:49:25 mountai Tue Jun 17 2008
! NVRAM config last updated at 14:49:27 mountai Tue Jun 17 2008
!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname HQ
!
boot-start-marker
boot-end-marker
!
enable secret 5 $1$IxmB$xIKVxlu3rdUOXR3SWjrDP.
enable password letmein
!
no aaa new-model
!
resource policy
!
memory-size iomem 10
clock timezone mountia -7
clock summer-time mountain recurring
!
!
ip cef
no ip dhcp use vrf connected
ip dhcp excluded-address 135.33.66.1 135.33.66.20
ip dhcp excluded-address 135.33.166.1 135.33.166.20
!
ip dhcp pool HQ-Voice
   network 135.33.66.0 255.255.255.0
   default-router 135.33.66.1
   option 150 ip 135.33.66.1
!
ip dhcp pool HQ-Data
   network 135.33.166.0 255.255.255.0
   default-router 135.33.166.1
   option 150 ip 135.33.166.1
!
!
!
!
voice-card 0
 dspfarm
!
```

```
class-map match-all VOICE
 match ip dscp ef
class-map match-all ROUTING
 match ip dscp cs6
class-map match-any CALL-SIGNALING
 match ip dscp cs3
 match ip dscp af31
!
!
policy-map BRANCH-LAN-EDGE-OUT
 class class-default
  set cos dscp
policy-map BRANCH-WAN-EDGE
 class VOICE
  priority percent 45
 class CALL-SIGNALING
  bandwidth percent 5
 class ROUTING
  bandwidth percent 3
 class class-default
  fair-queue
!
interface Loopback0
 ip address 1.1.1.2 255.255.255.255
!
interface GigabitEthernet0/0
 no ip address
 duplex auto
 speed 10
!
interface GigabitEthernet0/0.1
 description DVLAN SUBNET
 encapsulation dot1Q 200 native
 ip address 135.33.166.1 255.255.255.0
!
interface GigabitEthernet0/0.100
 description VVLAN SUBNET
 encapsulation dot1Q 100
 ip address 135.33.66.1 255.255.255.0
 service-policy output BRANCH-LAN-EDGE-OUT
!
interface FastEthernet0/1/0
!
interface FastEthernet0/1/1
!
interface FastEthernet0/1/2
!
interface FastEthernet0/1/3
!
interface Serial0/0/0
 ip address 172.16.22.2 255.255.255.248
 clock rate 2000000
 max-reserved-bandwidth 100
 service-policy output BRANCH-WAN-EDGE
 ip rtp priority 2000 16383 45
!
interface Vlan1
```

```
 no ip address
!
router ospf 1
 log-adjacency-changes
 network 1.1.1.2 0.0.0.0 area 0
 network 135.33.66.0 0.0.0.255 area 0
 network 135.33.166.0 0.0.0.255 area 0
 network 172.16.22.0 0.0.0.7 area 0
!
!
ip http server
no ip http secure-server
!
!
!
tftp-server flash:TERM41.7-0-3-0S.loads
!
control-plane
!
dial-peer cor custom
!
!
!
dial-peer voice 1 pots
 destination-pattern 3002
!
dial-peer voice 999 voip
 preference 1
 destination-pattern 1001
 session target ipv4:1.1.1.3
!
dial-peer voice 99 voip
 preference 1
 destination-pattern 2001
 session target ipv4:1.1.1.4
!
dial-peer voice 2 pots
 destination-pattern 3001
!
dial-peer voice 888 voip
 preference 1
 destination-pattern 1002
 session target ipv4:1.1.1.3
!
dial-peer voice 88 voip
 preference 1
 destination-pattern 2002
 session target ipv4:1.1.1.4
!
telephony-service
 no auto-reg-ephone
 load 7941 TERM41.7-0-3-0S.loads
 max-ephones 6
 max-dn 10
 ip source-address 135.33.66.1 port 2000
 max-conferences 8 gain -6
 transfer-system full-consult
```

```
 create cnf-files version-stamp 7960 Jun 17 2008 14:49:24
!
!
ephone-dn  1
 number 3001
 name A
!
!
ephone-dn  2
 number 3002
 name B
!
!
ephone  1
 mac-address 001B.D512.6C45
 type 7941
 button  1:1
!
ephone  2
 mac-address 001B.D52C.4046
 keepalive 45
 type 7941
 button  1:2
!
!
!
line con 0
line aux 0
line vty 0 4
 password letmein
 login
!
scheduler allocate 20000 1000
ntp clock-period 17179855
ntp server 1.1.1.3
!
end
```

Switch configuration

```
HQ-switch#sh run
Building configuration...

Current configuration : 3465 bytes
!
! Last configuration change at 19:11:42 mountai Sat Jun 21 2008
! NVRAM config last updated at 19:05:27 mountai Sat Jun 21 2008
!
version 12.2
no service pad
no service timestamps debug uptime
no service timestamps log uptime
no service password-encryption
!
hostname HQ-switch
!
```

```
enable secret 5 $1$xXH0$oLWHU9pjjSoRTS6HVb.5N/
!
no aaa new-model
clock timezone mountia -7
clock summer-time mountain recurring
switch 1 provision ws-c3750g-24ps
ip subnet-zero
!
no ip igmp snooping
!
mls qos map cos-dscp 0 8 16 24 32 46 48 56
mls qos
!
!
no file verify auto
spanning-tree mode pvst
spanning-tree extend system-id
!
vlan internal allocation policy ascending
!
class-map match-all Phone-Signaling
  match access-group name Ph-signal
class-map match-all VOICE
  match access-group name IP-VOICE
!
!
policy-map Phone-PC
  class VOICE
   set dscp ef
    police 128000 8000 exceed-action policed-dscp-transmit
  class Phone-Signaling
   set dscp cs3
    police 32000 8000 exceed-action policed-dscp-transmit
  class class-default
   set dscp default
    police 5000000 8000 exceed-action policed-dscp-transmit
!
!
!
interface GigabitEthernet1/0/1
 !
interface GigabitEthernet1/0/17
 description Monitor Port
 switchport mode dynamic desirable
!
interface GigabitEthernet1/0/18
!
interface GigabitEthernet1/0/19
!
interface GigabitEthernet1/0/20
 switchport trunk encapsulation dot1q
 switchport trunk native vlan 200
 switchport mode trunk
 switchport voice vlan 100
 service-policy input Phone-PC
 speed 10
 spanning-tree portfast
```

```
!
interface GigabitEthernet1/0/21
 !
interface GigabitEthernet1/0/22
!
interface GigabitEthernet1/0/23
 switchport trunk encapsulation dot1q
 switchport trunk native vlan 200
 switchport mode trunk
 switchport voice vlan 100
 service-policy input Phone-PC
 speed 10
 spanning-tree portfast
!
interface GigabitEthernet1/0/24
 switchport trunk encapsulation dot1q
 switchport trunk native vlan 200
 switchport mode trunk
 switchport voice vlan 100
 speed 10
 spanning-tree portfast
!
interface GigabitEthernet1/0/25
!
interface GigabitEthernet1/0/26
!
interface GigabitEthernet1/0/27
!
interface GigabitEthernet1/0/28
!
interface Vlan1
 no ip address
 shutdown
!
interface Vlan100
 ip address 135.33.66.2 255.255.255.0
 no ip route-cache
 no ip mroute-cache
!
interface Vlan200
 ip address 135.33.166.2 255.255.255.0
 no ip mroute-cache
!
ip classless
ip http server
!
!
ip access-list extended IP-VOICE
 permit udp any any range 16384 32767
ip access-list extended Ph-Signal
 permit tcp any any range 2000 2002
!
!
control-plane
!
!
line con 0
```

```
line vty 0 4
 no login
 length 0
line vty 5 15
 no login
!
!
monitor session 1 source interface Gi1/0/20 - 21 , Gi1/0/23
monitor session 1 destination interface Gi1/0/17
ntp clock-period 36028696
ntp server 1.1.1.3
end
```