QoS in VoIP

Rahul Singhai

Parijat Garg

Outline

- Introduction
- The VoIP Setting
- QoS Issues
- Service Models
- Techniques for QoS
- Voice Quality Monitoring
- Sample solution from industry
- Conclusion

Introduction

- Internet provides best-effort service
- Increasing bandwidths lead to new applications
- Applications have QoS demands
- Need mechanisms to provide QoS over best-effort network
- VoIP, streaming multimedia applications are new major applications

VoIP Setting

- All pervasive packet-switched network in place (Internet).
- Exploit this network for voice telephony; reduce dependency on circuit switched technology.
- Provide connectivity from Internet to PSTN network.

VoIP QoS Requirements

- Latency
  - High latency leads to problems in full-duplex communication. Critical for interactive voice applications.
- Jitter
  - Jitter buffers are used but effective only when small jitter values.
- Packet Loss
  - Slight loss (lesser than 1%) is tolerable but leads to deterioration of voice quality. Steps need to be taken for correction.

Service Models

- Integrated Service Model
  - Provides guaranteed service as well as controlled load service.
  - Does not scale well in core of network.
- Differentiated Service Model
  - Aggregates multiple flows into service classes.
  - Lesser control on delay / jitter for individual flows. Should be used with care.

Techniques for QoS

- Data Plane and Control Plane techniques.
- Control Plane
  - Route Selection.
  - Call Admission Control.
  - Resource Reservation.
- Data Plane
  - Queue management (WRED).
  - Loss Recovery.
  - Error Concealment.

QoS in Control Plane

- Route Selection
  - Constraint Based Routing to select routes with minimum / bounded delay.
  - Delay is additive constraint, so use techniques like Delay Scaling Algorithm.
  - Use MPLS, etc. for source based routing.
- Resource Reservation
  - Setup reservations using protocols like RSVP.
  - Use service models like IntServ and DiffServ.

QoS in Control Plane

- Call Admission Control
  - Based on failure of reservation under say RSVP protocol.
  - Based on preconfigured utilization thresholds of intermediate nodes.
  - Based on availability of route with certain delay thresholds.
  - Based on differentiation between high quality and low quality VoIP calls.

QoS in Data Plane

- Active Queue Management
  - Use techniques like WRED to control load on intermediate routers.
  - VoIP typically uses UDP so WRED does not throttle VoIP stream.
  - VoIP is less sensitive to packet loss (up to 1%) so dropping of some VoIP packets by WRED is acceptable.

QoS in Data Plane

- Jitter Buffers
  - Applications maintain Jitter Buffers to cover up for jitter in underlying best-effort network.
  - Out of order packet delivery can also be handled by same buffers.
- Loss Recovery
  - Packet retransmission not a good idea.
  - Latency becomes unacceptably high.

QoS in Data Plane

- Loss Recovery
  - Add redundancy to voice stream.
  - Two streams for same voice data, one with higher quality encoding and other with lower quality encoding.
- Error Concealment
  - Replay last correctly received packet.
  - Insert silence / background noise.
  - Interpolate from previous packet to next packet.

QoS in Data Plane

- Error Concealment (Sophisticated)
  - Use pattern matching to select some previous packet which matches current stream characteristics.
  - Use time scaling to *stretch* previous packet to cover time slot of missing packet.

Voice Quality Monitoring

- End-to-end Call quality
  - Voice quality
  - Call setup time
  - Call blocking rate
  - Call tear down time
- After Call setup
  - Most Important is voice quality.
  - Must be maintained for entire call duration.

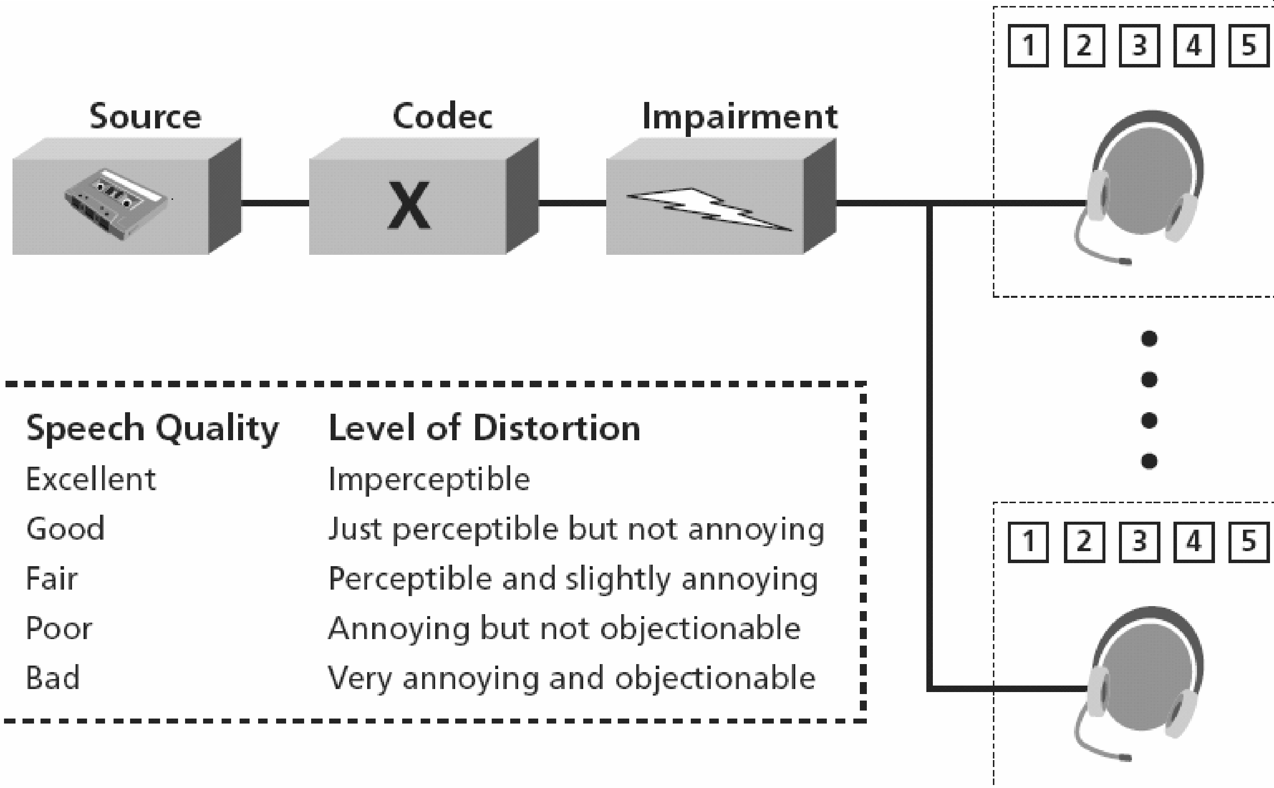
Voice quality measurement

- Subjective measures of voice quality
  - Mean opinion Score (MOS)
- Objective measures of voice quality
  - Perceptual Model
  - E-Model

Mean Opinion Score (MOS)

- ITU Recommendation P.800
- Human listeners score voice quality
- Score between 1 (bad) and 5 (excellent)
- Toll quality if mean score 4 or above
- PSTN Connections rated at 4.3
- Disadvantages
  - Expensive
  - Time Consuming
  - Inappropriate for general network measurements

Mean Opinion Score (MOS)



Rating	Speech Quality	Level of Distortion
5	Excellent	Imperceptible
4	Good	Just perceptible but not annoying
3	Fair	Perceptible and slightly annoying
2	Poor	Annoying but not objectionable
1	Bad	Very annoying and objectionable

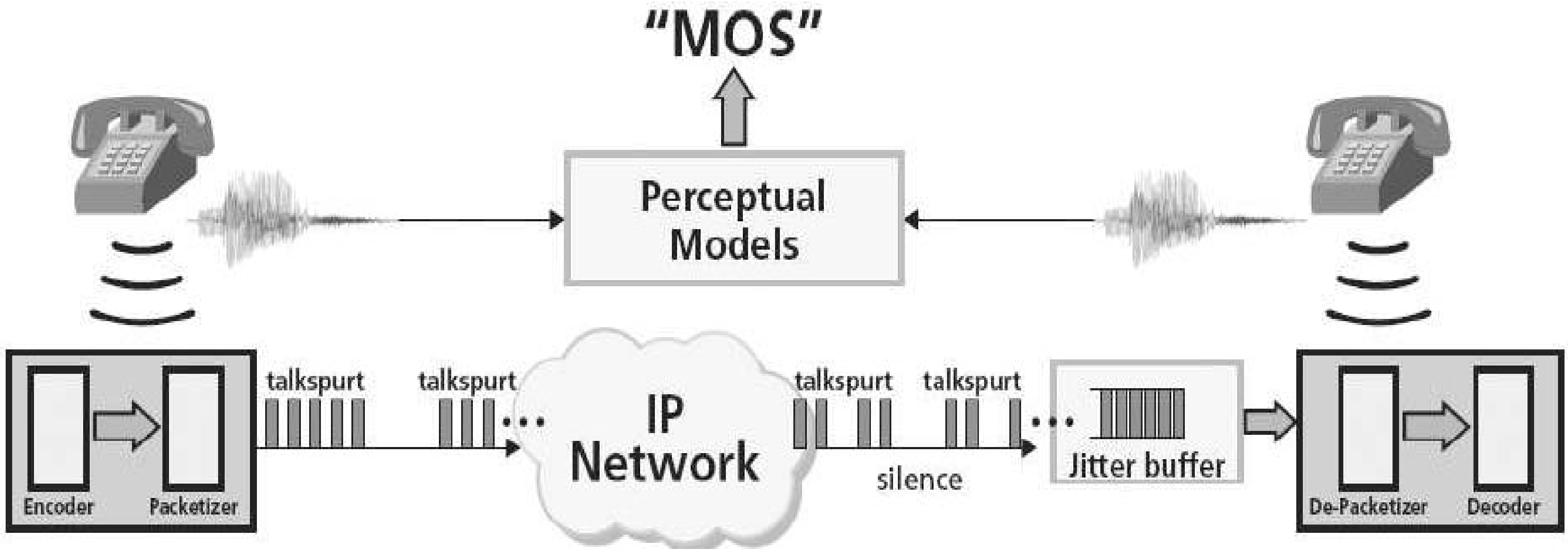
MOS of 4.0 = Toll Quality

Source : Reference 6

Perceptual Model

- Compares received speech signals to the sent ones in a psychoacoustic domain.
- Focus on one way distortion.
- Not scalable – synthetic calls.
- Do not show the causes of degradation of voice quality.
- Synthetic calls may even increase the load on network.
- Suitable for lab or prototype environments for capacity planning.

Perceptual Model



Source : Reference 6

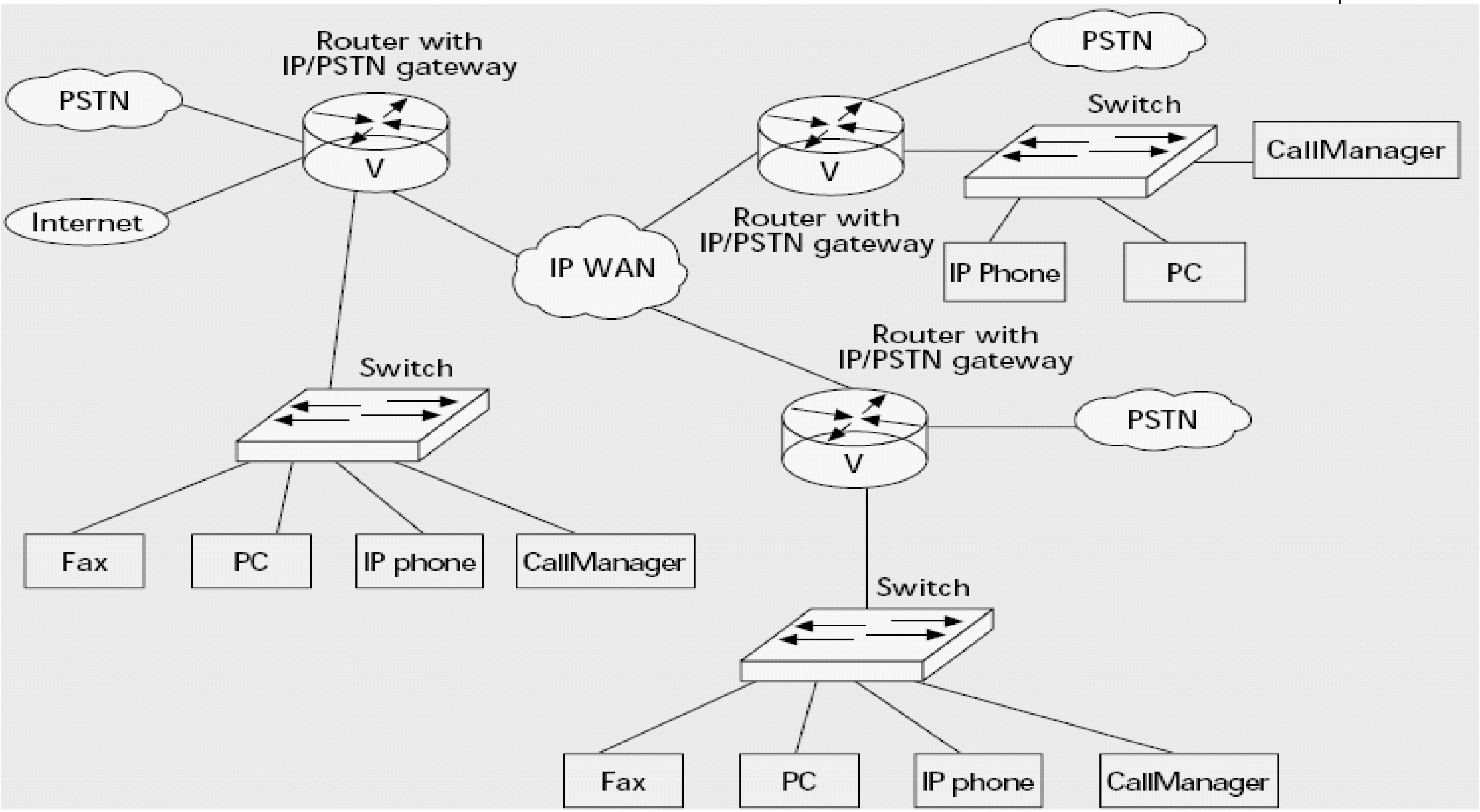
E-Model

- ITU Recommendation G.107
- Predicts rating done by “average user” when knowing characterizing transmission parameters.
- Calculates transmission rating factor R, using network impairment factors.
- Network impairment factors (like codecs, delay, and packet loss) obtained from extensive set of subjective experiments.
- R-value converted to MOS score.
- Appropriate for root-cause analysis of impairments and network segments.
- Scalable as it does not require speech samples between many pairs of nodes.

Cisco's Solution: Enterprise IP Telephony

- Includes
  - Switches
  - Routers
  - IP / PSTN Gateways
  - Desktop IP Phones
  - Call Managers

Cisco's Solution: Enterprise IP Telephony



Source : Reference 6

Cisco's Solution: Enterprise IP Telephony

- Call management software deployed at each remote site.
- Network supports multiple classes of services (CoSs)
- Provides guaranteed QoS to real-time communications.
- Packet classification and user policies applied at the edge of the network.
- Cisco IP phone sets the IPv4 ToS at ingress.
- QoS guarantees provided by two mechanisms:
  - Call manager
  - Priority queue mechanism

Cisco’s Solution: Enterprise IP Telephony

- Call manager
  - Equipped with RSVP
  - Priority queue mechanism.
    - Maintained in the core routers
    - Responsible for high-speed switching and transport and congestion avoidance (WRED).

Conclusion

- Introduction of QoS to IP networks
  - Affects all four performance measures (delay, jitter, frame loss and the out-of-order packets).
  - Service differentiation possible.
  - Many new services possible, among them will certainly be a high-quality IP telephony.
- QoS offered in two basic ways
  - Absolute QoS levels (absolute values of bandwidth, delay and other parameters are agreed)
    - Offered by technologies such as ATM and RSVP
  - Relative QoS levels (performance relative to priority class)
    - Offered by technologies such as TOS in IP networks or Precedence in Frame Relay networks.
- Other means to improve VoIP performance
  - Advanced jitter buffers that can adapt its length to the changing network conditions
  - Use of FEC
  - Loss concealment
- Voice Quality Measurement
  - Helpful for call admission control

References

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