End-to-end Voice Quality – The Impact of VoIP Cable Telephony in the Triple Play

Enabling Multiple System Operators (MSOs) to package video, data and voice services

The unique features of video, data and voice can pose a challenge for MSOs

This white paper addresses the Quality of Service (QoS) issues surrounding the Triple Play offer:

- The factors impacting voice quality
- Guidelines for providing end-to-end QoS in cable telephony
- How QoS can be monitored on an ongoing basis

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Introduction

Many Multiple System Operators (MSOs) are currently interested in offering VoIP cable telephony service by upgrading their network with new CableLabs® standards. DOCSIS® 1.0 was designed to offer high-speed data service without considering QoS because the data services are mostly non-real time applications. DOCSIS 1.1 included QoS, packet-fragmentation capabilities, and improved security to properly support the Voice over Internet Protocol (VoIP) and video applications. DOCSIS 2.0 was designed to accommodate the increased upstream bandwidth demand by adding advanced modulation techniques. The PacketCable specifications define interfaces for providing packet-based voice, video and other high-speed multimedia services over Hybrid Fiber Coax (HFC) cable systems utilizing the DOCSIS 1.1 protocol. The PacketCable 1.0 specifications were developed to implement a single-zone PacketCable solution for residential VoIP services. PacketCable 1.1 specifications define requirements for offering a primary line capable service using the PacketCable architecture. PacketCable 1.2 specifications define the functional components and interfaces necessary to allow communication between PacketCable 1.0 networks using an IP transport or backbone network.

When MSOs want to offer bundled service packages that include video, data, and voice services, adding voice services to the current MSO's network invites some serious challenges. Among them, providing Quality of Service (QoS) similar to that of the Public Switched Telephone Network (PSTN) will be critical to its success. The Dynamic QoS (DQoS) mechanism defined by the PacketCable project addresses part of this issue. However, voice quality of cable telephony is an end-to-end (phone-to-phone) matter including managed IP networks or the PSTN with various network equipment, protocols, and policies. The DQoS mechanism only specifies the "Access" portion (HFC network) of the PacketCable network, providing bandwidth to requesting applications on a per-flow basis. QoS options for IP access networks and backbone networks are left up to the MSO. In order to provide end-to-end QoS for VoIP cable telephony, it is imperative for MSOs to manage appropriate QoS levels across all the network segments.

In this paper, we present QoS options, analyze various impairment factors impacting voice quality in the PacketCable architecture, identify their sources, and discuss guidelines to provide end-to-end QoS for cable telephony. In addition, we discuss how QoS can be monitored on an ongoing basis and the proactive steps that can be taken to prevent congestion.

Multi-Service Environments

When MSOs want to offer bundled service packages including video, data, and voice services, one of the challenges the MSO confronts is meeting the performance requirements of these services simultaneously because these services have different traffic attributes. Table 1 lists these traffic attributes in terms of directionality, burstiness, time sensitivity, holding time, bandwidth, and performance metrics. It is clear that these attributes may be very different for each type of service. For example, voice service has symmetrical traffic with fairly modest burstiness and is sensitive with latency due to its real time characteristics. The holding time is relatively short and the bandwidth requirement per call is fairly small. The performance of voice service is sensitive to delay and the voice quality can tolerate some dropped packets within a certain limit of packet loss rates. On the other hand, in the case of data service, the traffic is vastly asymmetrical and highly bursty. Most data service is not very sensitive to latency, but cannot tolerate packet loss. The holding time is relatively long. Throughput is the major performance metric of data service, which depends on the available bandwidth. The table clearly illustrates that the four kinds of services have quite different traffic attributes from each other. If the MSO's network providing bundled service packages is designed for one particular type of service, it could be a rather poor network for another type, and vice versa.

	Data Service	Digital Video Distribution Service	Digital Video Communication Service	Voice Service
Directionality	Asymmetrical	Asymmetrical	Symmetrical	Symmetrical
Burstiness	High	Medium/High	Medium/High	Low
Time Sensitivity	Non-real time	Non-real time	Real time	Real time
Holding Time	Medium/Long	Long	Short/Medium	Short
Bandwidth Requirement	Medium	High	High	Low
Performance Metrics	Throughput	Throughput, Video quality	Delay, Video quality	Delay, Voice quality

Table 1. Traffic attributes of different services

Key VoIP Cable Telephony Issues for QoS

VoIP cable telephony is a real-time service and QoS is an important factor in the successful implementation of VoIP cable telephony. Low network latency and a predictable packet delivery mechanism are required to avoid jitter across the network. Let us look at key QoS issues in deploying the VoIP cable telephony service based on the PacketCable reference architecture, as shown in Figure 1.

The MSO must provide the end-users with the desired end-to-end call quality, which represents the performance of the VoIP cable telephony service the end users actually perceive. The end-to-end call quality consists of voice quality, call setup time, call blocking rate, call tear down time, etc. After a call is properly set up, voice quality is probably the most important characteristic for the entire call duration. The end-to-end voice quality must be maintained for the entire call duration.

Since voice packets are sent over connectionless IP networks in the VoIP cable telephony architecture, voice quality can be affected by various impairment factors such as delay, jitter, and packet loss. The MSO must properly configure the QoS parameters that can impact the performance of VoIP cable telephony service, including Embedded Multimedia Terminal Adapter (E-MTA), Cable Modem Termination System (CMTS), Media Gateway, and routers. Since these QoS parameters usually impact the performance of other services, careful planning is usually required to meet the performance requirements for VoIP cable telephony service as well as other services. The QoS implementation must be consistent across the network. If one of the network segments does not provide a proper QoS level, the end-to-end QoS would suffer. If the voice packets are to be transmitted over IP core network, an appropriate IP Transport Agreement (IPTA) must be negotiated to include the terms of the parameters impacting on the performance of VoIP cable telephony service.

To maintain network performance and deliver the desired QoS for VoIP cable telephony service, the MSO must have a proactive monitoring and management solution for the MSO's network. The MSO needs to be informed of potential service degradations and act on them before they impact the end-user.

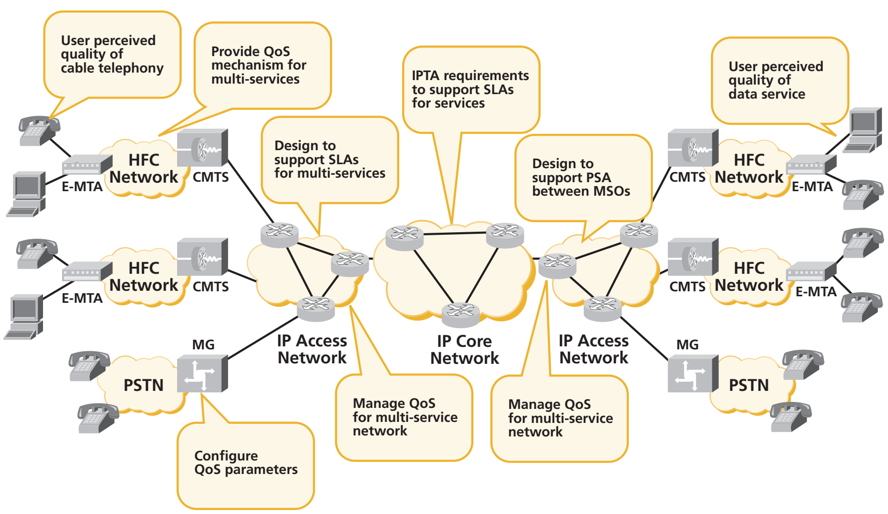


Figure 1. Key VoIP Cable Telephony Issues for QoS

QoS protocols

Implementing VoIP cable telephony requires adding real-time capabilities to the MSO's existing network and may require the MSO to install new devices conforming to the PacketCable specifications. There are a number of new network protocols related to QoS for VoIP cable telephony. DQoS and Resource Reservation Protocol (RSVP) are required for E-MTA and Standalone Multimedia Terminal Adapter (S-MTA), respectively in the HFC network based on DOCSIS 1.1 or 2.0.

DQoS provides a mechanism to allocate resources on the DOCSIS network for individual flows associated with each session of an application, per subscriber, on an authorized and authenticated basis.

RSVP is a signaling protocol to allow performance-sensitive applications to reserve bandwidth, so that the given applications can operate at a proper performance level. The ability to reserve bandwidth intends to prevent performance issues such as latency, jitter, and packet loss. In principle, with RSVP, users who wish to utilize a particular application can reserve bandwidth through the network.

Routers and switches in the IP access and core networks must be capable of understanding the new QoS protocols separating the voice from the data traffic. There has been a great deal of work done within the Internet Engineering Task Force (IETF) regarding QoS in the IP network. RSVP and Integrated Services (IntServ) specifications have gained some success in the access network, but little acceptance in the core network because of the scalability problem.

IntServ is a “circuit-oriented” model, which abandons the fundamental strength of the “packet-oriented connection-less” Internet Protocol (IP). Because all routers are required to maintain “state” information (amount of bandwidth and buffer currently reserved) per flow, the scalability is severely limited. Also, router cycles are wasted on reservation capture and processing. Per call bandwidth reservation with RSVP will not work with today's routers in an IP core network.

In the core network, a critical requirement for QoS mechanisms is scalability. It is, therefore, recommended that Differentiated Services (DiffServ) techniques be used in the core network. DiffServ defines ways of assigning specific service levels and priorities to IP traffic and provides various QoS mechanisms based on different classes of service. Traffic classification allows packets to be prioritized according to the needs of specific applications. Class of service networking ensures that traffic that is designated as high-priority always takes precedence over lower priority traffic. DiffServ is more scalable than RSVP because DiffServ doesn't require per-flow state in participating routers.

DiffServ, by itself, still doesn't solve the problem as voice calls can still overload the bandwidth allocated to the voice class. Thus some form of admission control will be required.

Network Impairments Affecting Voice Quality

End-users are extremely satisfied with the performance of voice services provided by the PSTN today: "five-nines" availability, excellent call quality, primary line, and guaranteed 911 or emergency service. The characteristics of VoIP cable telephony systems are different in many aspects from those of existing PSTN mainly due to the connection-less IP networks. Whereas the circuit-switched PSTN guarantees that sufficient bandwidth is reserved and available for the entire call duration, best-effort IP networks introduce significant new challenges to the delivery of real-time voice traffic. In the VoIP cable telephony architecture, voice quality can be affected by various impairment factors such as codecs, delay, jitter, and packet loss. As shown in Figure 2, such impairments are caused by the configuration of network equipment, network performance, and routing path of calls.

Delay causes two voice quality problems — echo and talker overlap. Echo is caused by the signal reflections of the speaker's voice. Echo becomes a significant problem when delay is greater than 50 milliseconds. Echo cancellation must be used to minimize the quality problem caused by echo. Talker overlap makes telephone conversation very difficult if one-way delay is greater than 250 milliseconds, so every effort must be made to minimize delay. The sources of delay in VoIP cable telephony are accumulation delay, codec algorithmic delay, processing delay, network delay, jitter buffer delay, and polling delay. Accumulation delay is caused by the need to collect a frame of voice samples to be processed by the voice coder. For example, G.711 in the PSTN does not need to have a frame and the accumulation delay is 0.125 milliseconds, a single sample time. However, G.728 and G.729 have 2.5 and 10 milliseconds of accumulation delay, respectively. In addition, it is typical to pack multiple frames into a single packet to minimize the impact of the packet header overheads.

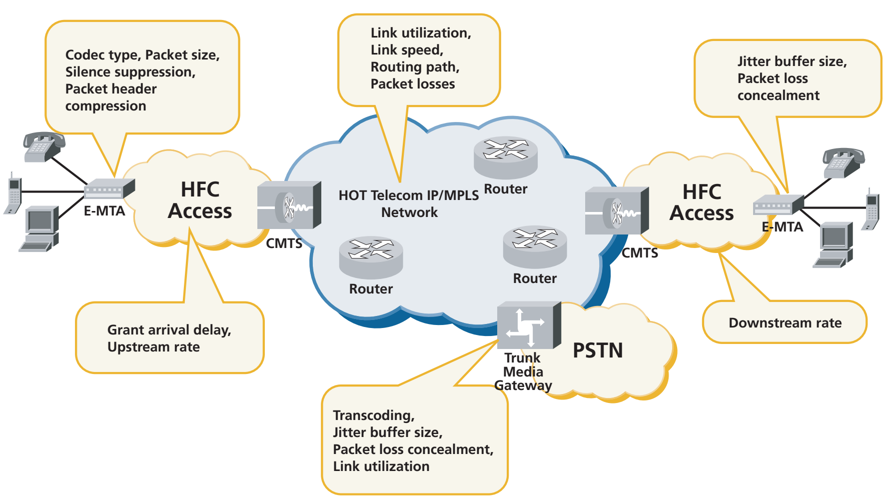


Figure 2. Causes of impairments affecting voice quality in VoIP cable telephony

Codec algorithmic delay is sometime called look-ahead delay. This delay is caused by the characteristics of a specific voice encoding algorithm. For example, G.729 has a codec algorithmic delay of 5 milliseconds. Processing delay is caused by the actual process of encoding and collecting samples into a packet for transmission. It is a function of the packet size, the processor execution time, and the type of algorithm used.

Network delay is caused by the physical medium used to move packets over the network. It is a function of the routing distance between the source and destination, the routing path, packet switching delay of routers, and queuing delay. In the terminating media gateway or E-MTA, a jitter buffer is used to smooth out the jitter and causes additional delay. Jitter is a variable delay in the delivery of each voice packet. Jitter is caused by the fact that packets do not all cross the network at the same speed due to routing path and congestion status. If a packet arrives later than the jitter buffer size, that packet is considered as dropped. VoIP cable telephony creates an additional delay in the HFC network because of the upstream bandwidth allocation of the Media Access Control (MAC) protocol. The Cable Modem Termination System (CMTS) polls the Network Interface (NI) at each customer location. Because the CMTS doesn't maintain a continuous connection with each NI, there is transmission delay while voice packets wait for the next poll. Therefore, it is important that VoIP cable telephony equipment minimize this delay by anticipating when the next poll will arrive — a process called grant synchronization — so that the packets are queued up and ready to go.

In best effort IP networks, voice packets are treated exactly like data packets. Under peak loads and congestion, voice packets will be dropped at the same rate as data packets. The data packets, however, are not time sensitive and dropped packets can be corrected through retransmission using Transmission Control Protocol (TCP). Lost voice packets cannot be handled in the same manner due to the real time nature of voice.

QoS Features of Network Equipment

There are several QoS features used in network equipment such as media gateways and routers to provide QoS in the IP networks, VoIP cable telephony as well as other services. It is important to understand these QoS features and use them properly.

Packet classification and marking

To guarantee bandwidth for voice packets, a network device must be able to identify voice packets in all the IP traffic flowing through it. This identification and grouping process is called packet classification. Media gateways and MTAs must be able to use various match criteria to place traffic into a certain number of classes. Packet marking is the process of setting the Type of Service (TOS) bits or Differentiated Service Code Point (DSCP) bits in the IP header. With Multi-Protocol Label Switching (MPLS) in the core the edge router must also translate these bits into EXP bits in the MPLS shim header. Media gateways and MTAs must be able to configure packet marking based on various match criteria. The three most significant bits of the TOS byte are called the IP precedence bits and the six most significant bits of the TOS byte are called the DSCP, which can be used to define DiffServ classes. The packet marking can be used to represent the packet's delay priority and drop priority.

Priority queuing

After all traffic has been marked by QoS classes based on their QoS requirements, a network device needs to provide bandwidth guarantees and priority servicing through an intelligent output queuing mechanism. A priority queue is required for VoIP cable telephony because it is delay sensitive. This priority queue must have the capability to provide priority to certain classes and to provide guaranteed minimum bandwidth for other classes. During periods of congestion, the priority queue is policed at the configured rate so that the priority traffic does not consume all the available bandwidth and other classes avoid a situation of bandwidth starvation.

Echo cancellation

Echo is present even in a conventional PSTN network. However, it is not a problem for local calls because delay is less than 50 milliseconds and the echo is masked by the normal side tone that every telephone generates. Echo becomes a problem in a VoIP cable telephony network where the delay is usually greater than 50 milliseconds. Thus, echo cancellation techniques must be used. The International Telecommunication Union (ITU) standards G.165 and G.168 define performance requirements for echo cancellers. Echo is generated toward the packet network from the telephone network. The echo canceller compares the voice data received from the packet network with voice data being transmitted to the packet network. The echo from the telephone network is removed by a digital filter on the transmit path into the packet network.

Media gateways must be able to support echo cancellation to provide near-end echo cancellation on a per-connection basis.

Voice activity detection

With voice activity detection, only audible speech is transmitted over the network and silence is not transmitted. When the voice activity detection feature is enabled, the voice quality will be slightly degraded, but the connection will require less bandwidth.

Jitter buffer setting

The jitter buffer delay is the amount of time that elapses between the time that a voice packet is received at the jitter buffer and the time that it is played out to the codec. Media gateways and MTAs must be able to configure the jitter buffer delay parameter in either a fixed or adaptive jitter buffer to improve voice quality by reducing jitter. If the jitter buffer delay becomes longer to reduce packet loss, the network delay would become longer. So, the jitter buffer delay parameter must be configured properly. In fixed mode, the jitter buffer delay set at the beginning of a call by the jitter buffer in the gateway is applied throughout the call. As a general rule, if there is excessive breakup of voice due to jitter with the jitter buffer delay settings, increase the jitter buffer delay parameter. If the network jitter is small, decrease jitter buffer delay for a smaller overall delay. In adaptive mode, the jitter buffer grows and shrinks to compensate for jitter and to keep voice packets playing out smoothly, within the maximum and minimum limits. The maximum limit establishes the highest value to which the adaptive delay will be set. The minimum limit is the low-end threshold for the delay of incoming packets by the adaptive jitter buffer.

Call admission control

The call admission control feature provides the ability to support resource-based call admission control processes. These resources include system resources such as CPU, memory, and call volume, and interface resources such as call volume. If system resources are not available to admit the call, two kinds of actions are provided: system denial (which busy outs all of a T1 or E1) or per call denial (which disconnects, hairpins, or plays a message or tone). If the interface-based resource is not available to admit the call, the call is dropped from the session protocol. Another type of call admission control feature is based on the measurement of network congestion status. The measurement-based call admission control algorithm derives the network congestion information between the source and destination gateways through call quality reports or direct network measurements. If the network congestion status is above a preset threshold value, the path is assumed to be congested and a call is blocked. The performance of the measurement-based algorithm depends on the measurement interval. This CAC algorithm guarantees a certain level of voice quality to admitted VoIP calls even when the network is congested.

Traffic policing based on traffic class

Traffic policing can be deployed to ensure that a packet, or data source, adheres to a stipulated contract and to determine the QoS to render the packet. Both policing and shaping mechanisms use the traffic descriptor for a packet stream—indicated by the classification of the packet—to ensure adherence and service. Policers and shapers usually identify traffic descriptor violations in an identical manner. They usually differ, however, in the way they respond to violations, for example:

A policer typically drops traffic. A shaper typically delays excess traffic using a buffer, or queueing mechanism, to hold packets and shape the flow when the data rate of the source is higher than expected. Traffic shaping and policing can work in tandem. For example, a good traffic shaping scheme should make it easy for nodes inside the network to detect misbehaving flows. This activity is sometimes called policing the traffic of the flow.

Congestion avoidance techniques such as WRED

Weighted Random Early Detection (WRED) makes early detection of congestion possible and is a means of avoiding Transmission Control Protocol (TCP) synchronization thus improving system throughput. WRED can selectively discard lower priority traffic when the router begins to experience congestion and provide differentiated performance characteristics for different classes of service. It also protects against global synchronization. Global synchronization occurs as waves of TCP congestion crest, only to be followed by periods of time during which the transmission link is not used to capacity. For these reasons, WRED is useful on any output interface or router where congestion is expected to occur. WRED is usually implemented at the core routers of a network. Edge routers or voice gateways assign IP precedences to packets as the packets enter the network. With WRED, core routers then use these precedences to determine how to treat different types of traffic. WRED provides separate thresholds and weights for different IP precedences, enabling the network to provide different qualities of service, in regard to packet dropping, for different types of traffic. Standard traffic may be dropped more frequently than prioritized traffic during periods of congestion.

Voice Quality Measurements

Before adding new services or network components to the existing network infrastructure, a service provider needs to determine the potential impact of the changes. Usually the focus is on evaluating the network performance or component itself. This may leave out end-user experience on the services. Emphasis on end-user experience means improving end-to-end call quality for VoIP cable telephony service, reducing response time for data service rather than just network uptime or performance.

End-to-end call quality consists of voice quality, call setup time, call blocking rate, call tear down time, etc. After a call is properly set up, voice quality maintained for the entire call duration is probably the most important characteristic. The end-to-end voice quality must be maintained for the entire call duration.

As described before, in the VoIP cable telephony architecture, voice quality can be affected by various impairment factors such as codecs, delay, jitter, and packet loss. Such impairments are caused by the configuration of network equipment, network performance, and the routing path of calls. Among them, network performance must be monitored continuously due to its dynamic changes. A well-managed network is necessary to provide the desired level of VoIP cable telephony service. If the voice quality were below the desired level, it would be necessary to perform root-cause analysis based on the measurements of network performance. Once the causes of the degraded voice quality are diagnosed, the problems must be fixed and it is important to ensure the solution really fixed the problems and did not cause any new problems.

There are several standard methods to measure voice quality – International Telecommunication Union (ITU) Recommendations P.800 (MOS), P.861 (PSQM), P.862 (PESQ), and G.107 (E-Model). The Mean Opinion Score (MOS) defines a method to derive a mean opinion score of voice quality after collecting scores between 1 (bad) and 5 (excellent) from human listeners. This is a form of subjective testing because human listeners are involved. In subjective testing, subjects (human listeners) are required to classify the perceived quality into categories (excellent, good, fair, poor, bad). In each subjective experiment, the MOS scores may differ, even for the same condition, depending on the design of the experiment, the range of conditions included in the study, etc. Since human listeners are involved, such subjective testing is very expensive and time-consuming. Therefore, it cannot be performed in real time.

Perceptual Models

In the mid 1990s, the ITU began to standardize objective speech quality measures designed to estimate subjective voice quality. A robust objective speech quality measure should correlate well with subjective speech quality. There are two types of objective speech quality measures: perceptual models and the E-Model. Perceptual models estimate the voice quality by comparing the received speech signal to the sent speech signal in a psychoacoustic domain. The Perceptual Speech Quality Measure (PSQM) and Perceptual Evaluation of Speech Quality (PESQ) belong to perceptual models. The perceptual models focus on the effects of one-way speech distortion and they do not consider other impairments related to two-way interaction, e.g. delay. The perceptual models are not scalable because they need to inject the speech samples at one end point and receive them at another end point in order to measure voice quality between two end points. If the voice quality becomes degraded, the perceptual models could not show the causes of degradations. These measures only get a snapshot of system performance by monitoring synthetic calls or average calls, not "real" calls. Additionally, by adding synthetic calls on the network, these measures can exacerbate conditions being tested by increasing load on the network. This tends to make the perceptual models more suitable for lab or prototype environments for capacity planning type activities.

E-Model

The E-Model (ITU-T G.107) is a tool for predicting how an "average user" would rate the voice quality of a phone call with known characterizing transmission parameters. It estimates the user satisfaction of a narrowband, handset conversation, as perceived by the listener. The E-Model calculates the transmission rating factor R, using the network impairment factors, which were obtained after an extensive set of subjective experiments. Typical network impairment factors used in the VoIP cable telephony are codecs, delay, and packet loss. After computing the R-value based on the impairment factors, the R-value is converted into an MOS score. Since the E-Model is based on the measurements of impairments, it is appropriate for root-cause analysis in terms of impairment factors as well as network segments, and can be easily incorporated with the Network Management System (NMS). The E-Model is also scalable because it does not require the speech samples to estimate the voice quality.

The E-Model consists of several models that relate specific impairment parameters and their interactions to end-to-end performance. The total end-to-end performance, taking into account all factors, is estimated using the Impairment Factor method. The equation for the transmission rating factor R is:

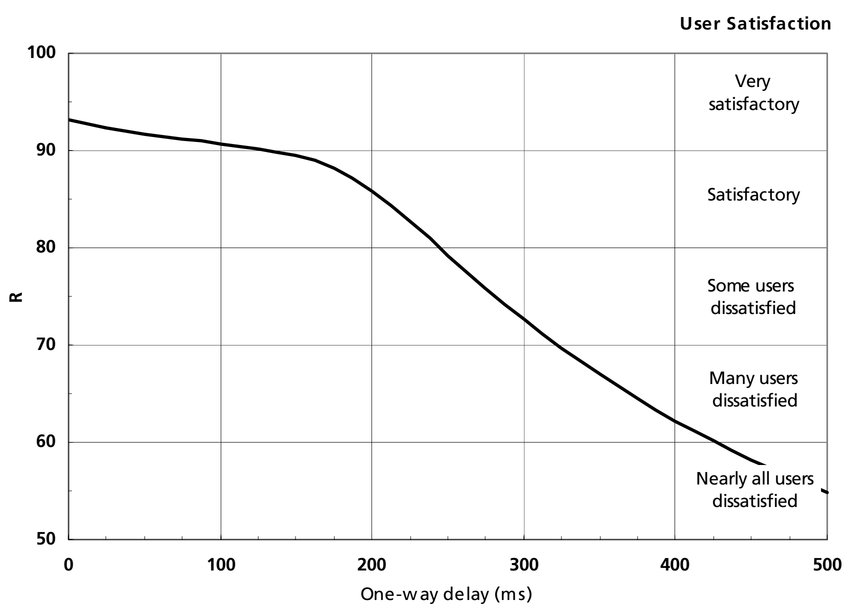
$$R = R_0 - I_s - I_d - I_e + A$$

Where,

R_0	the basic signal-to-noise ratio based on sender and receiver loudness ratings and the circuit and room noise
I_s	the sum of real-time or simultaneous speech transmission impairments, e.g. loudness levels, sidetone and PCM quantizing distortion
I_d	the sum of delay impairments relative to the speech signal, e.g., talker echo, listener echo and absolute delay
I_e	the equipment impairment factor for special equipment, e.g., low bit-rate coding (determined subjectively for each codec and for each % packet loss and documented in ITU-T Recommendation G.113)
A	the advantage factor adds to the total and improves the R-value for new services.

Assuming that echo is properly controlled by an echo cancellation module, one can review the impairments of the E-Model in terms of delay, codec impairments, and packet loss.

The curve in Figure 3 plots the transmission rating factor R versus one-way delay for the reference connection. The right-hand side of Figure 3 includes the "User Satisfaction" scale for reference. The reference connection curve uses the E-Model default value (93.19) for all parameters except the variable delay. This gives the best possible performance for a narrowband handset conversation, over the range of one-way delay, and therefore will be used as the "relative reference". Based on this curve, if the delay is the only VoIP impairments, a "very satisfactory" rating requires a one-way delay less than about 140 ms.



Delay (ms)	0	50	100	150	200	250	300	350	400
R	93.19	91.74	90.65	89.53	85.79	79.17	72.66	67.02	62.24

Figure 3. Delay Impairment of Reference Connection

The E-Model is flexible to deal with the impairments introduced by speech codecs and packet loss via the equipment impairment factor (I_e). I_e values for several codecs are listed in ITU-T Recommendation G.113 and Table 2 lists a few of them. The I_e values in Table 2 were determined in subjective experiments with ideal software implementation of the codecs; the performance provided by commercial codecs may vary.

Codec Type	Codec	Bit Rate (Kbps)	Ie Value
PCM	G.711	64	0
ADPCM	G.726 G.726 G.726	40 32 24	2 7 25
LD-CELP	G.728	16	7
CS-ACELP	G.729-A + VAD	8	11
RPE-LTP	GSM-Full Rate	13	20
VCELP	GSM-Half Rate	5.6	23
ACELP	GSM-EFR	12.2	5
MP-MLQ	G.723.1	5.3	19
MP-MLQ	G.723.1	6.3	15

Table 2. Speech codecs and their I_e values

The equipment impairment factors (I_e) for G.711, G.729A, and G.723.1 codecs under conditions of packet loss are listed in Table 3. Packet Loss Concealment (PLC) algorithm will be strongly recommended when G.711 codec is used under conditions of packet loss.

Packet Loss (%)	G.711 without PLC	G.711 + PLC Random Packet Loss (10ms speech packet)	G.711 + PLC Bursty Packet Loss (10ms speech packet)	G.729A + VAD 8 Kbps (2 speech frames/packet)	G.723.1 + VAD 6.3 Kbps (1 speech frame/packet)
0	0	0	0	11	15
1	25	5	5	15	19
2	35	7	7	19	24
3	45	10	10	23	27
4	–	–	–	26	32
5	55	15	30	–	–

Table 3. Packet loss impairment and their I_e values

Guidelines for providing end-to-end voice quality for VoIP Cable Telephony

As a new service, the MSO must plan, design, deploy, operate, and manage the VoIP cable telephony solution to meet the desired performance objectives. In order to provide end-to-end voice quality for VoIP cable telephony, it is crucial for the MSO to perform the following two components: network assessment and planning, and network and service management.

Network Assessment and Planning

When the MSOs want to add VoIP cable telephony service, they must assess if the existing network is ready to deploy VoIP cable telephony service and not degrade the performance of existing services. Such an assessment investigates the ability of the existing network to support the additional traffic, QoS protocols, QoS features, and new network equipment caused by a new service. It is important to include the impact of a new service on the existing service.

Next, the MSOs need to perform planning for bandwidth requirement, network capacity, configurations of network equipment, Packet Service Agreement (PSA) with other MSOs, Internet Protocol Transport Agreement (IPSA) with IP service provider to meet the desired performance of VoIP cable telephony service.

Network and Service Management

To maintain network performance and deliver the desired QoS for VoIP cable telephony service, the MSO must have a proactive monitoring and management solution for the MSO's network. The MSO need to be informed of potential service degradations and act on them before they impact the end-user. To achieve this, network performance must be monitored continuously due to its dynamic changes. Since a well-managed network is necessary to provide the desired level of VoIP cable telephony service, end-to-end voice quality must be monitored in real-time. It is important to recognize that a voice call is comprised of two unidirectional flows, and that the routing path of each direction may differ. It is imperative to obtain measurements for each direction of the flow, versus a round-trip measurement in order to isolate problems. The voice quality is determined by the worst quality of two unidirectional flows. If the voice quality were below the desired level, it would be necessary to perform root-cause analysis based on the measurement of network performance. Once the causes of the degraded voice quality are diagnosed, the problems must be fixed and it is important to ensure the solution really fixed the problems and did not cause any new problems.

After the VoIP cable telephony solution is deployed, the network architecture and its capacity change as the traffic demands vary over time. It is necessary for MSOs to respond to the quickly evolving network environment efficiently. Network optimization could be required to improve utilization efficiency of network resources while keeping the desired performance levels of the services the MSOs' network provides. If the desired performance for some services is not met with the given network resources, the network capacity must be increased to maintain the performance of those services.

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v1 04/04

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