Application-based Quality of Service

For IP Video Conferencing

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Introduction

IP video communications and voice over IP (VoIP) applications differ from traditional data applications in that they are real-time applications, and as such, require higher bandwidth than traditional data applications. Real-time applications, especially video communications and VoIP applications, also require more stringent packet handling than non-real-time (traditional IP-based) applications. This, in turn, necessitates more network management resources to assure predictable traffic flow. Video/VoIP communications can require significant bandwidth and must receive service with minimal delay, jitter, and loss.

Traditional discussions of network-based IP Quality of Service (NQoS) discuss only the networking facilities related to Quality of Experience (QoE) of IP video/voice interchange, such as IP precedence, differentiated services, Intserv (RSVP), router queue management, and so on. Although traditional discussions of QoS only include the networking facilities related to QoE, not all networks are NQoS-capable or NQoS-enabled. Therefore, application-based QoS (AQoS) services often must provide whatever QoE is available for voice/video sessions. In fact, *best effort* QoS is the most widely deployed and scalable NQoS implementation available today. The intent of this white paper is to review AQoS.

The quality of an end user's experience is the true litmus test of a proper video/voice deployment. Only by understanding both the application and network facilities for QoE can you absolutely ensure the highest quality user experience. This white paper discusses the breadth of AQoS within H.323 video/VoIP applications and highlights several AQoS solutions provided by Polycom's IPriority™ solution suite. Hopefully, this will provide you with a greater understanding of how important AQoS mechanisms are in the overall scheme of QoE.

Information Sources

This white paper has derived some of its information from existing publications, such as other white papers and RFCs. In addition, some terminology and descriptions directly pertain to a specific publication. In these cases, a number, in brackets, follows the term or description. The number identifies the specific publication that contains the information about that term or description. The publications are numbered and listed in “References” on page 21. For example, this white paper contains the term Application-based Quality of Service [1] (AQoS). The [1] identifies the number of the publication that contains the information about Application-based Quality of Service. In this case, the AQoS information is derived from *IP Telephony with H.323: Architectures for Unified Networks and Integrated Services*.

Application-based Quality of Service [1] (AQoS)

Application-based Quality of Service (AQoS) relates to the facilities embedded within an application that preserve the quality of its intended use. For H.323 video/VoIP applications, AQoS relates to two main areas:

- Call signaling
- Terminal handling of media flows, both video and audio.

There are several opportunities for providing AQoS services within these two areas.

AQoS and the Application Flow Process

Understanding the application’s flow process is a simple way of separating the functions of AQoS. Table 1 on page 3 shows the generic process of application flow from a sender's acquisition, analog to digital conversion of media, and packetization to the receiver’s acquisition, de-packetization, digital to analog conversion, and play back.

Table 1: H.323 Application Process [1] From Media Acquisition (at Sender) to Transmission Over the Network to Media Playback (at Receiver)

Step	Description
1	Input to camera and microphone (media) is digitized and buffered at sender
2	Media retrieved from buffer and entered into Codec
3	Compression of media
4	Output of compression is sent to RTP for packetization
5	RTP output sent to UDP and IP layers for packetization
6	Application marks for NQoS signaling
7	IP datagram is transmitted
8	Routing occurs (injection of propagation-serialization delay based on queuing)
9	H.245 via RTP measures loss, jitter, latency and instructs application to react
10	Media is de-packetized at receiver’s buffer
11	Codec retrieves from buffer and adjusts delay to account for loss and jitter
12	Output is played

AQoS and Signaling

There are three main signaling mechanisms in H.323:

- H.225-Registration, Admission, and Status (RAS)
  RAS, setup, and connect are all signaling processes relating to H.323. The topics in this paper include what the ITU H.323 standard says about RAS, H.225, and H.245. It is recommended that you understand the responsibility of each process and the method for its signaling. The standard is clear on areas of recommended AQoS to overcome the trade-offs made between speed and reliability.
- H.225-Q.931
- H.245

Each signaling mechanism provides areas for AQoS. The process of call setup follows, as described in ITU H.225 [4]:

1. Mandatory admission request is sent on the unreliable (UDP) RAS channel.
2. Initial setup message is sent on a reliable (TCP) channel transport address (IP address).
3. Call setup sequence commences, based on Q.931 operations.
4. The sequence is complete when the terminal receives a reliable transport address (IP address) in the connect message. This address is used to send H.245 control messages.
5. Additional channels for audio, video, and data can now be established, based on the result of H.245 channel capability procedures.

The Real-time Transport Protocol (RTP) and Real-time Transport Control Protocol (RTCP) are also involved. RTP and RTCP provide mechanisms for terminal-to-terminal connection control services in point-to-point and multi-point calls.

RAS, Setup, Connect

RAS, setup, and connect are all processes provided for in the H.323 standard. The following information provides descriptions from the ITU H.323 standard for:

- Registration, Admission, and Status (RAS) Channel [1]
- H.225 [4]
- H.245 [4]

The descriptions explain the responsibility of each process and the method for its signaling. This information is provided to clarify the areas where AQoS can be used, keeping in mind the balance between speed and reliability of transfer.

Note: In the following topics, many processes are underlined and in italics. These processes are described in the signaling process of H.323 applications and are potential tools for optimization related to delay and reliability.

Registration, Admission, and Status (RAS) Channel [1]. The RAS channel is used to carry messages used in the *Gatekeeper discovery* and endpoint *registration* processes, which associate an endpoint's alias address with its Call Signaling Channel Transport Address (IP address). The RAS channel is an *unreliable channel*. Because the RAS messages are transmitted on an unreliable channel, H.225.0 recommends *timeouts and retry counts* for various messages. An endpoint or gatekeeper that cannot respond to a request within the specified timeout can use the *request in progress* (RIP) message to indicate that it is still processing the request. An endpoint or gatekeeper receiving the RIP shall *reset its timeout timer* and retry counter.

H.225 [4]. H.225.0 is used to set up and tear down a call. The delay and accuracy of these processes are areas of potential AQoS. H.323 terminals are capable of sending audio and video using RTP via *unreliable channels* to minimize delay. *Error concealment* or other *recovery action* can be used to overcome lost packets. In general, audio/video packets are not retransmitted because doing so would result in excessive delay in the LAN environment.

H.245 [4]. The H.245 recommendation provides a number of different services. Procedures are defined to allow the following:

- *Exchange of audiovisual and data capabilities* (capabilities exchange). Used to request the transmission of a particular audiovisual and data mode.
- *Manage the logical channels* used to transport the audiovisual and data information.
  Used to:
  - Establish which terminal is the master terminal and which is the slave terminal for the purposes of managing logical channels
  - *Carry various control and indication signals*
  - Control the bit rate of individual logical channels and the whole multiplex
  - *Measure the round-trip delay* from one terminal to the other and back.

RTP/RTCP

This section describes Real-time Transport Protocol (RTP) and Real-time Transport Control Protocol (RTCP). *RTP controls the media streams* for real-time applications, such as voice and video. *RTCP is a companion protocol to H.245,* which gathers statistical information about quality of network and codec performance. Receiver reports (RR) are used to provide feedback from receiver to sender.

Many of the signaling processes use an *unreliable channel*, which is a channel without error checking or feedback mechanisms. Because of this, some of the processes can be, by definition, unreliable. For example, TCP is reliable, whereas UDP is unreliable.

Note: In the preceding topics, many processes are underlined and in italics. These processes are described in the signaling process of H.323 applications and are potential tools for optimization related to delay and reliability.

Several techniques you can apply to the processes to provide AQoS are explained below.

AQoS for Signaling Delay

Signaling delay directly affects the speed at which an application can perform or react to signaling messages. Using some of the following techniques to optimize the processing delay of both terminal and gatekeepers can limit the latency of signaling:

- To speed up the RAS process, the gatekeeper can pre-grant address requests (ARQs) to terminals
- If security is used, delay is incurred. Use mechanisms to speed authentication, such as caching.
- H.245 capabilities exchange optimization can decrease the delay of call establishment and also predefine a higher quality for capabilities negotiated
- Review the process of registration. For example, Unicast is faster than Multicast.
- The delay incurred during the call admission phase can be improved with optimization, for example, by gatekeepers performing faster database queries.

AQoS for Signaling Reliability

To improve signaling reliability, RAS can be performed in a reliable channel, for example, using TCP instead of UDP. Although this is not recommended, this illustrates the point of trading off speed for reliability.

Other examples are:

- Retransmitting RAS messages after a minimal timeout to improve reliability
- Sending duplicate packets in rapid succession to double the odds of the RAS message's reception. This is known as *diversity.*

AQoS and Media Handling

You can use H.245 to provide AQoS services through its packet loss measuring and notification services. Packet loss, inter-arrival packet jitter, and one-way delay (latency) are all measurable. In addition, their existence can be reported to the application. When no NQoS exists, H.245 is the only QoS facility capable of reacting to network loss, jitter, and latency.

There are several techniques Polycom uses to leverage these facilities:

- Dynamic jitter buffers to compensate for inter-arrival packet jitter
- Dynamic bandwidth allocation (DBA)
- Polycom Video Error Concealment (PVEC).

AQoS Summary

The following is a short list of the AQoS services that can be obtained through signaling and media handling:

- Gatekeeper discovery/registration
- End-point registration
- Call admission
- Capabilities exchange
- Bandwidth negotiation
- Media handling
- Call disengagement.

These services are areas of potential AQoS and optimization. The time it takes for each service to complete and the accuracy and reliability of the service directly affects QoE.

Polycom Video Error Concealment

Historically, video communications over packet-switched networks has been plagued by packet loss. Typically, when a video packet is lost, the video communications participants experience distracting frozen video, blocky video, or worse. Polycom Video Error Concealment (PVEC) works to recognize errors due to packet loss and produces video that conceals the effects of that packet loss. This section contains a brief background about low bit rate video communications, followed by a high-level description of the PVEC algorithm.

Background

On a television, the moving image is made up of still video frames that are updated several times per second. Each video frame is made up of many small points called *pixels*. In North America, using NTSC, there are 720 x 480 active pixels on a display. The display is completely refreshed every thirtieth of a second.[1] If we assume 2 bytes per pixel and multiply the 2 bytes by the number of pixels that are painted on the screen every second, we get 2x720x480x30 bytes/second = 20,736,000 bytes/second = 165,888,000 bits/second.

1. Technically, each frame is made up of two interlaced fields.

Because of the cost of bandwidth for video transmission, the video signal must be compressed. For example, an NTSC video signal must be compressed by a ratio of 518 to 1 to be transmitted at a commonly used call rate of 384 Kbps, where 320 Kbps are used for video. Commonly used video compression techniques, capable of producing quality video at low data rates, include H.261, H.263, and MPEG-4.

The process for communicating low-rate video over packet-switched networks is summarized in the following list.

Video Transmitter

1. Image acquisition.
2. Macroblock decomposition of video frame.
3. Codec produces bits for each macroblock, one at a time.
4. Bits are put into packets.
5. Packets are sent onto the network.

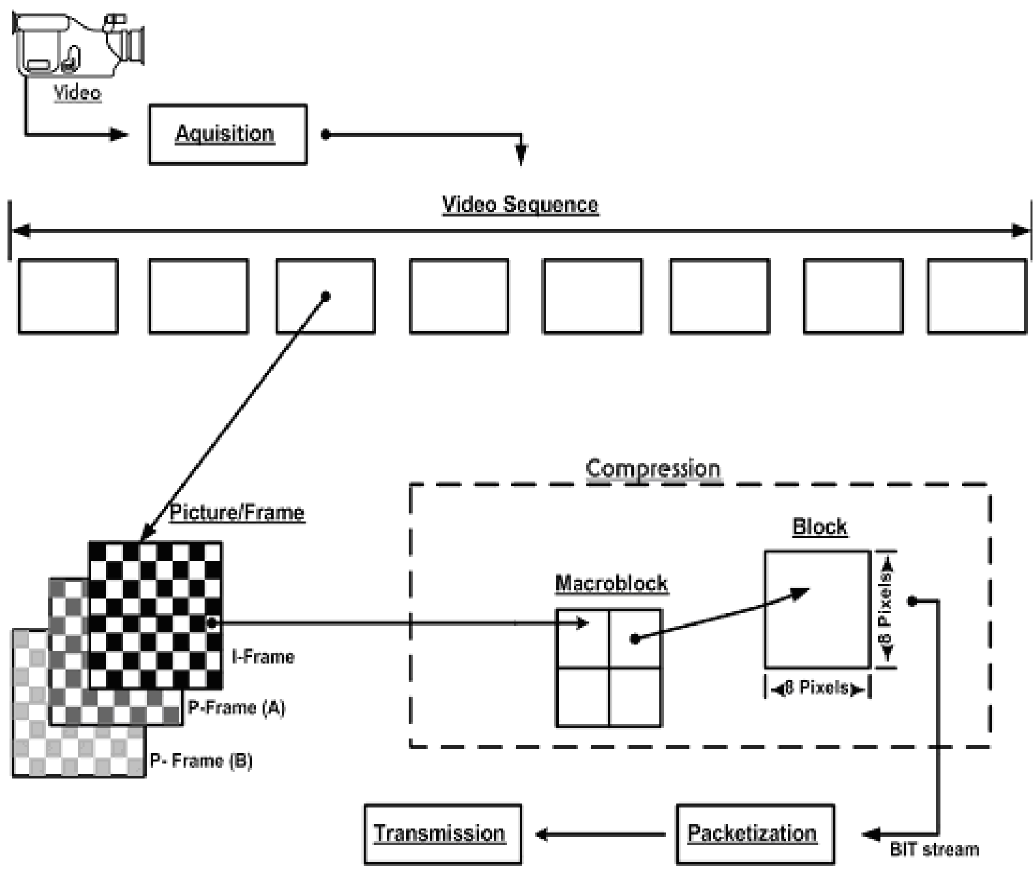
Video Receiver

6. Packets arrive at the receiver.
7. Bits from packets go to the video decoder.
8. A video frame is produced by the decoder.
9. The decoded video frame is displayed.

Figure 1 on page 9 highlights the video transmitter (items 1 to 5 in the above list).

Unfortunately, in practice, packet-switched networks lose packets. When packet loss occurs, a typical decoder will experience an error and video will freeze for a second or two while the video encoder and decoder work to clear the error.

Figure 1: Low-Rate Video Communications



PVEC Algorithm

An industry first, the patent-pending Polycom Video Error Concealment (PVEC) technology maintains the quality of video communications over an error-prone IP network by immediately recognizing and taking action on video packet loss. Before PVEC, video packet loss rates as low as one percent could render video communications unusable. For an illustration, see the lower half of Figure 2, “Packet Loss With and Without PVEC,” on page 11. PVEC, in conjunction with Polycom's new error resilient G.722.1 audio, makes video communications possible even when packet loss rates reach 10%.

PVEC is part of Polycom's IPriority initiative, which is designed to provide robust Quality of Service over IP networks. In Figure 2 on page 11, the standard (non-PVEC) encoder will packetize the macro blocks serially, in a raster-scan order. Consequently, packet loss results in the loss of large regions in the video frame because data for a contiguous group of macroblocks is lost. PVEC, on the other hand, scrambles the macroblocks before packetization. Macroblocks that are adjacent to one another are placed in different packets. Therefore, if packets are lost during transmission, PVEC needs to only compensate for the isolated missing macroblocks.

Polycom video terminal technologies can dynamically respond to packet loss over IP networks. This takes into account the impact of all other applications running on the network, not just the video communications traffic. Polycom terminal technologies can additionally compensate for packet loss rates of up to 10%. This is revolutionary in the fact that most other competitive offerings work well only to about 1% of packet loss.

The following is the PVEC sequence:

PVEC at the Transmitter:

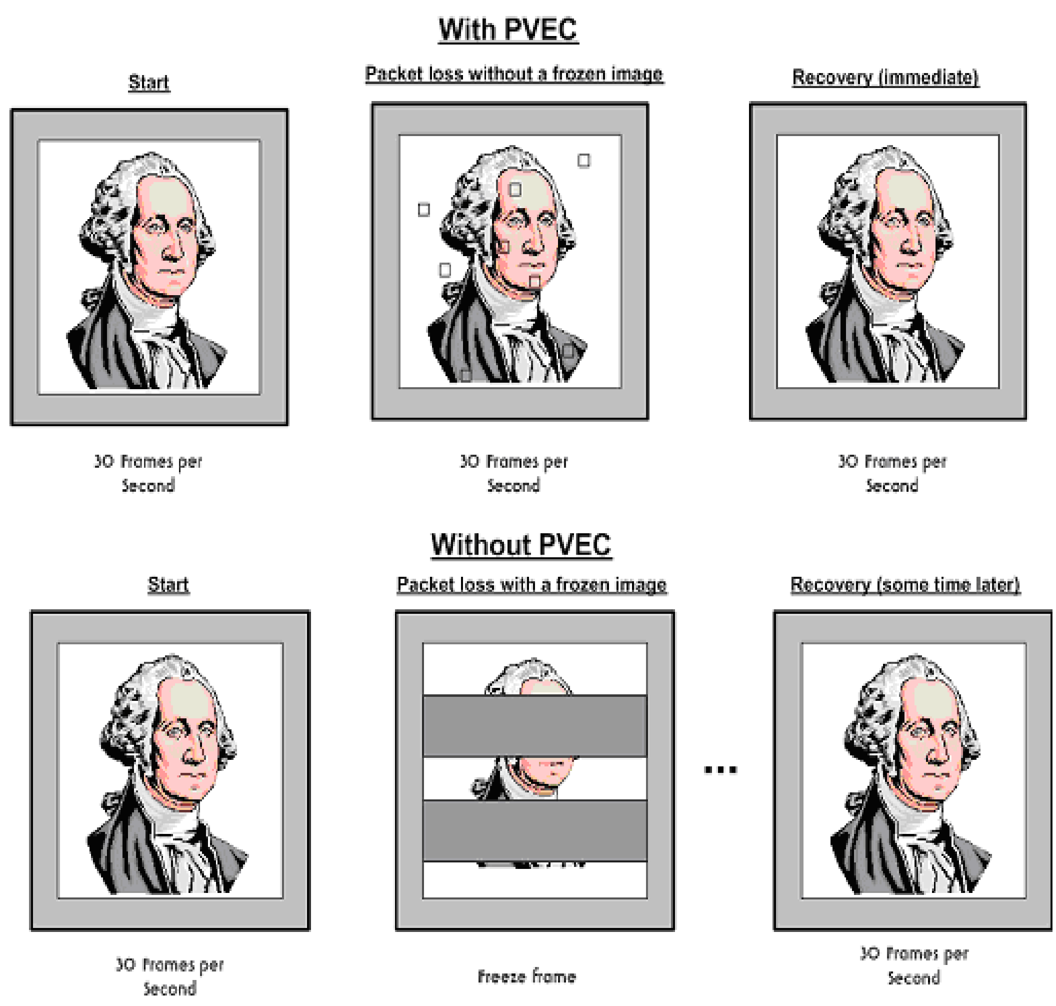
1. Acquire the image.
2. Scramble the macroblocks.
3. Code the macroblocks independently, using a codec almost identical to H.263. The difference is that H.263 macroblocks are not coded independently.
4. Packetize the scrambled macroblocks.

PVEC at the Receiver:

1. De-packetize the macroblocks.
2. De-scramble the macroblocks.
3. Identify the lost macroblocks.
4. Decode the macroblock data.
5. Conceal the lost macroblocks by using information from neighboring blocks that were received. Basically, PVEC completes the parts of the video that were lost, based on the parts that were not lost.
6. Display the video.

A key difference between standard video implementations and PVEC is the scrambled macroblock order. The scrambling helps to boost concealment performance by leveraging the fact that, as a result of the scrambling process, a lost packet results in the loss of an isolated macroblock. Consequently, the surviving macroblocks are used to develop an excellent guess about the content of the lost macroblocks, thereby yielding excellent error concealment.

Figure 2: Packet Loss With and Without PVEC



PVEC Summary

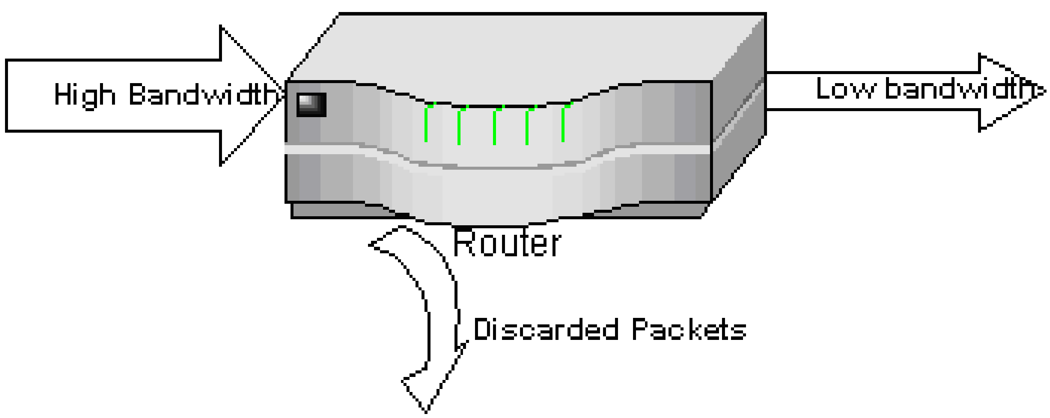
In networks without end-to-end QoS, Polycom equipment is capable of compensating for bandwidth and congestion-induced packet loss. Even in networks with QoS implementations (IP precedence, differentiated services), packet loss can and does occur. Polycom is uniquely prepared to provide innovative, industry-leading video communications technologies. These technologies complement standards-based NQoS functionality and provide for the highest level of Quality of Experience.

Polycom Video Error Correction

Dynamic Bandwidth Allocation [3]

Traditionally, video communications applications have transported compressed audio and video data streams over nearly lossless circuit switched networks, such as POTS and ISDN phone lines. These networks typically have error rates in the 10e-6% to 10e-7% range. Packet-switched networks have losses from 0% to 100%. Typically, loss occurs in routers when transferring packets from a higher bandwidth communication line to a lower bandwidth communication line as shown in Figure 3.

Figure 3: Packet Loss in a Router



Routers are often connected to networks that have constant bit rates so that no loss occurs as long as the incoming traffic on the high bandwidth line is less than the low bandwidth network's maximum sustainable bit rate. Once the traffic exceeds the low bandwidth network's maximum rate, packet loss occurs proportional to the amount of traffic over that rate.

When trying to conduct video communications over a lossy packet-switched network, what starts out as intelligible audio becomes laden with intermittent gaps of silence and the video exhibits a wide range of undesirable artifacts, including video freeze frame. Typically, video is more sensitive to data loss than audio. For example, a loss of 5% to 15% of the transmitted audio data can sometimes be tolerated (for example, cell phone calls), whereas a loss of even 1% or 2% for video data can make a video communications difficult to follow.

Two approaches can be used to reduce data loss in video communications applications on packet-switched networks. The first approach includes network-based Quality of Service (NQoS) methods, such as IP precedence, Diffserv, RSVP, or MPLS. These network-based techniques prioritize audio and video data over non-real time traffic, for example, HTTP, FTP, and so on.

The second approach is application-based Quality of Service (AQoS), in this case, end point application. The second approach requires the video communications endpoint to modify the data streams to accommodate for the loss in the network. This section focuses on the second approach and introduces an algorithm called dynamic bandwidth allocation (DBA), which dynamically adjusts the bandwidth during video communications in order to eliminate packet loss.

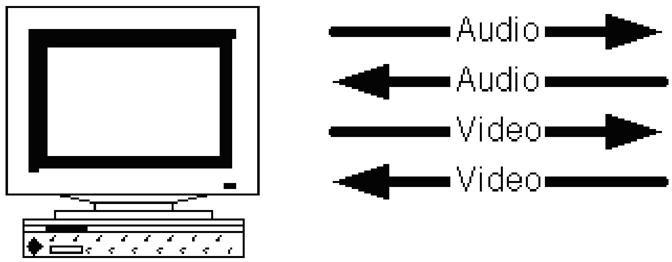
The techniques described here attempt to alleviate packet loss under the two most common packet loss conditions:

- **Condition A:** a constant packet loss condition that can occur when a high bandwidth network is connected to a low bandwidth network
- **Condition B:** a short burst of packet loss caused by intermittent heavy usage of the network.

Packet-Switched Video Communications

Protocols, such as H.323 or SIP, can be used for video communications over packet-switched networks. These protocols define the signaling (control) portion of the communication and typically send the compressed audio and video information as four separate streams using the IETF standard Real Time Protocol (RTP). This is shown in Figure 4, “Packet-based Audio and Video Streams,” on page 14.

Figure 4: Packet-based Audio and Video Streams



Not only are the streams separate, but control of the streams is also separate. Typical video controls might include algorithm selection and specification of bit rate, resolution, frame rate, and so on. Because most audio algorithms have a fixed data rate, control of audio streams is usually limited to algorithm selection.

Measuring Packet Loss Using RTP/RTCP

For each individual unidirectional audio/video RTP stream, there exists a bi-directional RTCP stream that carries control and status information for the associated RTP stream. This is shown in Figure 5 on page 15.

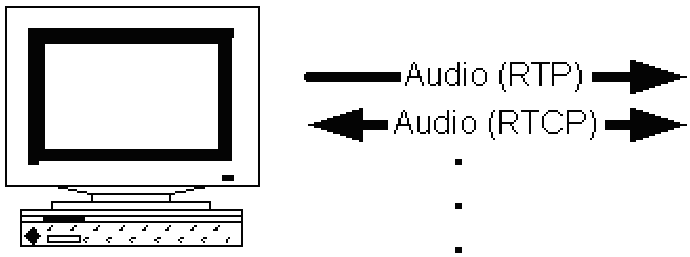
RTCP packets are sent approximately every 5 seconds by the receiver to the sender (Receiver Report packet) and every 5 seconds by the sender to the receiver (Sender Report packet).

Information in a report packet includes:

- Total packet loss since the RTP session started
- Current jitter and fraction packet loss, as defined in IETF standard RFC 1889.

The fraction packet loss is the fraction of packets lost since the last Receiver Report was issued.

Figure 5: RTCP Streams



Adjusting the Data Rate in Video Communications

The DBA algorithm dynamically adjusts the video bit rate used during video communications in order to eliminate packet loss. No rate adjustments are made for audio because audio is typically sent in a fixed rate data stream.

The dynamic bandwidth allocation algorithm's main features include:

- Rate adjustments may be asymmetric. That is, the receive video data rate is adjusted separately from the transmit video rate.
- Receive packet loss is determined by local processing of incoming RTP packets. The same information is used to generate RTCP Receiver Report packets.
- Transmit packet loss rate is determined by the RTCP Receiver Report from the remote endpoint.
- The same algorithm is used to adjust the local transmit and remote receive data rates when transmit packet loss occurs.
- The video data rate is rapidly adjusted to attain an acceptable operating point, that is, a data rate at which packet loss occurs at an acceptable level. In time, the algorithm attempts to achieve the rate originally specified in a way designed to minimize the disturbance to the ongoing video communications.

DBA Examples

Two examples of the DBA algorithm are presented in this section. The first example focuses on the behavior of DBA in the context of Condition A, a constant packet loss condition that results when a high bandwidth connection is connected to a low bandwidth connection. A second example demonstrates the behavior of DBA in the context of Condition B, where a short burst of packet loss is caused by intermittent heavy usage of the network.

Example: DBA in the Context of Condition A

In this example, Condition A is encountered, which is a constant packet loss condition that results when a high bandwidth connection is connected to a low bandwidth connection.

Table 2 explains the example in detail. Specifically, an endpoint places a 384 Kbps call that connects to an endpoint in which the network is capable of supporting a maximum data rate of 128 Kbps.

Figure 6 on page 17 shows the data rate compared to time.

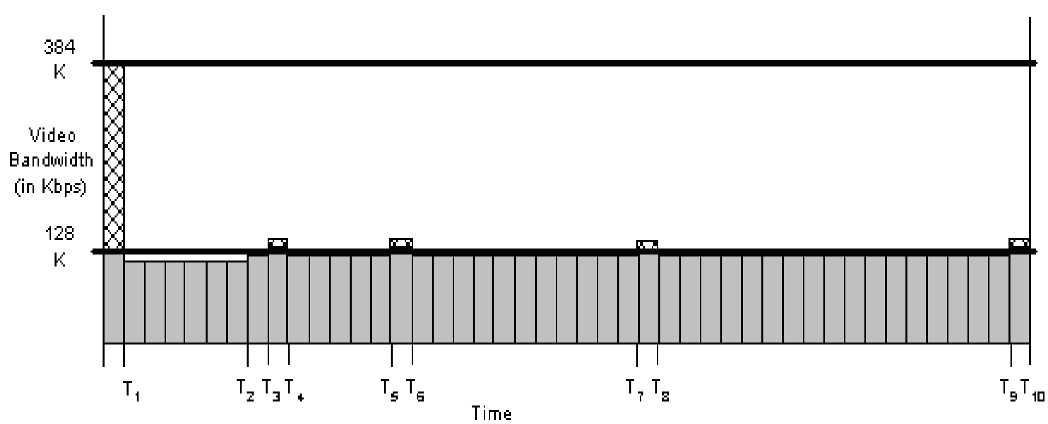
Note: For the sake of simplicity, the audio bandwidth is ignored in this example.

Table 2: Example Effects of the DBA Algorithm on the Video Bit Rate When the Network is Constrained by a Fixed Bandwidth Connection (Condition A)

Time	Bit Rate	Packet Loss	Description
0	384 Kbps	66%	Initial bit rate. Video is frozen due to heavy loss.
$\mathbf{T_1}$	111 Kbps	0%	Video decreased due to packet loss. No loss measured at this rate. New rate saved as “Last Known Good Rate.”
$\mathbf{T_2}$	122 Kbps	0%	No packet loss measured for $\mathbf{t_m}$ seconds. Video rate increased. No packet loss measured at this rate. New rate saved as “Last Known Good Rate.”
$\mathbf{T_3}$	134 Kbps	4%	No packet loss measured for $\mathbf{t_m}$ seconds. Video rate increased. Rate increase leads to packet loss. Revert to “Last Known Good Rate.”
$\mathbf{T_4}$	122 Kbps	0%	No packet loss. Wait $\mathbf{t_m}$ seconds.

Time	Bit Rate	Packet Loss	Description
T_5	134 Kbps	4%	No packet loss measured for **t_m** seconds. Video rate increased. Rate increase leads to packet loss. Revert to "Last Known Good Rate."
T_6	122 Kbps	0%	No packet loss. Wait **$2*t_m$** seconds.
T_7	134 Kbps	4%	No packet loss measured for **$2*t_m$** seconds. Video rate increased. Rate increase leads to packet loss. Revert to "Last Known Good Rate."
T_8	122 Kbps	0%	No packet loss. Wait **$3*t_m$** seconds.
T_9	134 Kbps	4%	No packet loss measured for **$3*t_m$** seconds. Video rate increased. Rate increase leads to packet loss. Revert to "Last Known Good Rate."
T_{10}	122 Kbps	0%	No packet loss. Wait **$4*t_m$** seconds.

Figure 6: Data Rate Vs Time for Example DBA in the Context of Condition A



Example: DBA in the Context of Condition B

In this example, Condition B, where a short burst of packet loss is caused by intermittent heavy usage of the network, is encountered.

Table 3 explains the example in detail. Specifically, an endpoint places a 384 Kbps call that connects properly, but experiences a burst of packet loss during the video communications.

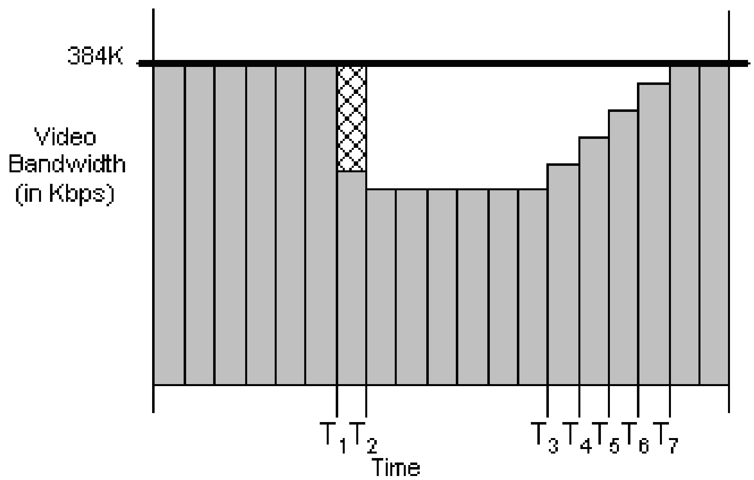
Figure 7 on page 19 shows the data rate compared to time.

Note: For the sake of simplicity, the audio bandwidth is ignored in this example.

Table 3: Example Effects of the DBA Algorithm on the Video Bit Rate in the Presence of a Short Burst of Packet Loss (Condition B)

Time	Bit Rate	Packet Loss	Description
0	384 Kbps	0%	Initial bit rate.
$\mathbf{T_1}$	384 Kbps	30%	Burst of packet loss.
$\mathbf{T_2}$	249 Kbps	0%	Video bit rate decreased by the RateReductionFactor = 0.65 due to packet loss.
$\mathbf{T_3}$	274 Kbps	0%	No packet loss measured for $\mathbf{t_m}$ seconds. Video rate increased by the RateExpansionFactor = 1.10.
$\mathbf{T_4}$	302 Kbps	0%	No packet loss measured for $\mathbf{t_m}$ seconds. Video rate increased by the RateExpansionFactor = 1.10.
$\mathbf{T_5}$	332 Kbps	0%	No packet loss measured for $\mathbf{t_m}$ seconds. Video rate increased by the RateExpansionFactor = 1.10.
$\mathbf{T_6}$	365 Kbps	0%	No packet loss measured for $\mathbf{t_m}$ seconds. Video rate increased by the RateExpansionFactor = 1.10.
$\mathbf{T_7}$	384 Kbps	0%	Video rate is restored to its original value.

Figure 7: Data Rate Vs Time for Example DBA in the Context of Condition B



DBA and PVEC work independently of each other. However, PVEC will derive benefit from DBA reducing overall packet loss through down speeding. The combination of DBA and PVEC represents the most advanced capabilities for preserving video quality over IP networks. Both of these technologies are patent-pending and exclusive to Polycom.

Polycom Video Error Concealment and Correction Summary

Polycom video terminal technologies can respond dynamically to packet loss over IP networks. This takes into account the impact of all other applications running on the network, not just the video communications traffic. Polycom terminal technologies can additionally compensate for packet loss within the IP network in the range of 1-10%. This is revolutionary in the fact that most other competitive offerings work only within a 1-2% range of packet loss. In networks without end-to-end QoS, Polycom is equipped to compensate for bandwidth and congestion-induced packet loss. Even in networks with QoS implementations (IP precedence, differentiated services), packet loss can and does occur. For the highest quality user experience in video application deployments, Polycom is uniquely prepared to provide innovative, industry-leading technologies designed to complement standards-based networking QoS functionality and provide for the highest level of Quality of Experience.

Terms and Acronyms

This white paper uses the following terms and their acronyms.

Table 4:Terms and Acronyms

Acronym	Term
AQoS	Application-based Quality of Service
ARQ	Address request
DBA	Dynamic bandwidth allocation
NQoS	Network-based Quality of Service
PVEC	Polycom Video Error Concealment
QoE	Quality of Experience
QoS	Quality of Service
RAS	Registration, Admission, and Status
RIP	Request in progress
RR	Receiver report
RTP	Real-time Transport Protocol
RTCP	Real-time Transport Control Protocol
TCP	Transmission Control Protocol
UDP	User Datagram Protocol

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   - Systems and Terminal Equipment for Audiovisual Services
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