Towards building H.323-aware 3G wireless systems: H.323 Control Loops and Application adaptation to wireless Link conditions

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Abstract

Third Generation Mobile Systems are required to support Multimedia applications with high Quality of Service (QoS) requirements. QoS provisioning is, therefore, a major issue in both 3G wireless networks and packet based multimedia systems such as H.323-based systems. Both 3G wireless networks and H323-based multimedia applications provide control loops. The former use low level open and closed loops driven by measurements of transmission power and other internal system resources in order to maintain link quality, while the latter use out off band Real Time Control Protocol (RTCP) and H.245 signaling to cope with fluctuating network conditions and congestion. However, very much like signaling in TCP (Transport Control Protocol), H.323 signaling was originally tailored for fixed networks. Understanding the interactions and the cross-induced effects due to both types of control loops can lead to wireless link optimization, and to building H.323-aware/multimedia-aware 3G wireless systems capable of playing an active role in the end to end H.323 signaling by intelligently forwarding/changing/spoofing control packet reports.

This paper first investigates control loops in H.323-based multimedia applications, identifies wireless network events/conditions that might trigger them, and suggests some solutions to adapt the different control loops identified to a wireless link.

Keywords: H.323, Control loops, Quality of Service, Wireless Systems, Multimedia.

1. Introduction

The traditional Internet was originally built to support best effort data traffic with no QoS guarantees. A great deal of research has been devoted to upgrade the Internet to support Multimedia transmission at both the content level and the delivery level. However, most of the adopted mechanisms are typically tailored for fixed networks, while it is not yet investigated how they would perform on wireless links. In a Wireless environment, other problems emerge, such as handoff, power control, interference, etc. Wireless links are characterized by their great transient behavior. Unlike wired links, they are subject to both topology changes and to high error rates due to the characteristics of the air interface (interference of the propagating signals). These problems result in varying bandwidth and varying error rates. The former is most of the time low while the latter is most probably high [9,1]. These inherent characteristics of the wireless link make guaranteeing and provisioning Quality of Service of multimedia applications in such an environment a challenging task. Middleware control loops need to be added on top of existing wireless control loops like power control to account for the multimedia component. In addition, adaptation based on the feedback of the receiver at the level of the wireless receivers should be supported.

IP-based multimedia applications use specific control mechanisms to adapt to the network conditions. For example, in H.323[1] [7], a standard for multimedia applications in packet-based networks, congestion is measured by means of feedback from RTP/RTCP (the real time protocol and the real time control protocol) [3] and H.245 Control messages [4]. Accordingly, video, data, audio, and control messages are switched off in this order. Surveying the main higher lever control loops in H.323 multimedia applications and identifying the different problems in wireless environments that might trigger them would provide the basis for: **1.** Learning in order to predict the appropriate actions to be taken once faced with one of the Network problems due to Multimedia transmission. **2.** Being able to place intelligence at the edge of the wireless network and hence being able to support intelligent multimedia aware Middleware in a wireless environment.

The rest of this paper is organized as follows. In section 2, a survey of the H.323 transport level and codec level control loops will be given. Section 3 discusses the problems that H.323 control loops might encounter in wireless links and proposes a solution to adapt these control loops to the wireless link. Conclusions and future works follow in section 4.

2. QoS issues in H.323 Network

The relevant QoS parameters for multimedia flows in all packet-based networks are:
1. Bandwidth, **2.** Maximum Delay, **3.** Jitter or Variation in delay, and **4.** Packet loss. In H.323-based multimedia networks, QoS is addressed at two levels: the transport level and the codec level. At the transport level, H.323 relies mainly on RTCP reports and on H.245 control messages for QoS monitoring and receivers' feedback. At the codec level, different mechanisms are used for applications adaptation to Network behavior. Codec adaptation is triggered by transport level signaling messages.

2.1 Transport Level-QoS monitoring in H.323

2.1.1 RTCP QoS Monitoring

In an RTP session [3], receivers and coders periodically send RTCP packets to convey feedback on quality of data delivery and information of membership. RFC 1889 defines five RTCP

[1] ITU has been recently developing an extension to H.323 for mobility support in H.323 Annex H.

packet types to carry control information: **RR** (Receiver Report) for reception quality feedback, **SR** (Sender Reports) for inter-media synchronization, **SDES** (Source Description items) for describing the different sources, **BYE** to end the participation, and **APP**, which is application dependent.

RTCP control packets are interpreted by higher-level applications. They are used to assess QoS and to trigger accordingly, adaptation mechanisms at the codec level. Two main responses have been determined in [6]:
Short-term responses, which result from temporary delays and not congestion such as:

- ✓ Reduction of the frame rate for a short period of time.
- ✓ Reduction of the packet rate by switching to the optional mode by which both audio and video are mixed or multiplexed in one RTP packet.
- ✓ Use of MB fragmentation of the video stream to reduce the packet rate.

Long term responses, which are due to network congestion such as:

- ✓ Reduction of media bit rate either through an indication to the encoder or through the use of a rate reducer function in the gateway. The signaling is done through the use of H.245 Flow-control commands [4].
- ✓ Disabling media of less importance. For example disabling video and maintaining T.120 traffic, which is more important.
- ✓ Using the adaptive busy signal, which is returned by the sender to the receiver as an indication of congestion. It is signaled via the cause clause in the Q.931 Release Complete message [5].

2.1.2 H.245 QoS Control messages

H.245 procedures are used for capability exchange, channel negotiation, and flow control. The procedures used for QoS management and control are part of the Round Trip Delay Determination and Commands and Indications procedures. Table 1 shows the different commands and Indications used for QoS Control and their descriptions.

H.245 Commands and Indications	Description
VideoFastUpdatePicture	Commands the video encoder to perform fast-update, the earliest possible.
VideoFastUpdateGOB	Commands the video encoder to perform a fast update of one or more GOBs. It is used with both H.261 and H.263 codecs.
VideoTemporalSpatialTradeOff	Commands the video encoder to change its trade-off between temporal and spatial resolution. A value of 0 commands a high spatial resolution and a value of 31 commands a high frame rate.
VideoSendSyncEveryGOB	Commands the video encoder to use sync for every GOB until the command videoSendSyncEveryGOBCancel is received. Used in H.263.
VideoFastUpdateMB	Commands the video encoder to perform a fast update of one or more MBs.
DoOneProgression	Commands the video encoder to begin producing a progressive refinement sequence. This mode is kept until an acceptable level of fidelity is reached.
DoOneIndependentProgression	In this mode, the video encoder produces video data consisting of one Intra picture followed by a sequence of 0 or more frames of refinement of the quality of the same picture. This mode is kept until an acceptable fidelity level is reached.
DoContinuousIndependentProgressions	This command is similar to the precious one. The only difference is that we have here continuous refinement of the video.
Jitter Indication	This is used to indicate the estimated amount of jitter of a logical channel. The video encoder can use this information to restrict the video bit-rate.
Flow Control Command	This command is used to specify the upper bit rate of a specific logical channel or the entire multiplex. Requesting a channel to transmit at a rate lower than the lowest rate allowed result in stopping the media transmission.

Table 1: H.245 QoS Control Messages

2.2 Codec Level-QoS monitoring in H.323

Video Codec support is optional in H.323 terminals. When it exists, the support of the H.261 encoding is mandatory. H.263 and H.263+ are newer versions of H.261 that have been developed to support the newer picture formats. H.323 supports additionally other video codecs and other picture formats, such as MPEG2.

2.2.1 H.261 Codec

H.261 [3] supports two encoding schemes: Inter-frame coding and intra-frame coding. In the first one, only the difference between the subsequent frames is sent. In Intra-frame coding the 8x8 blocks are coded independently. H.261 encoding is packetized with RTP then carried over UDP, which has poor resistance to loss. To decode the GOB data, one needs to receive the information present in the picture Header. Similarly, decoding the MBs requires the information present in the GOB headers. So the receiver needs to receive all the information in order to be able to decode properly and play back the audio and video streams. This hierarchical encoding hence makes H.261 more sensitive to packet loss. To counter this problem, different measures can be taken [6]:

- First, each packet has a state Header information from the frame Header and the GOB Header to allow the packets to be decoded and processed independently in case of packet loss.
- The MB is taken as a level of the fragmentation i.e., packets must start and finish on an MB boundary. This practice reduces the sending rate and packets overhead.

To alleviate loss problems due to Inter-frame coding in H.261several mechanisms are proposed in recommendation H.261 [6]:

- Using INTRA-frame coding and MB level conditional replenishment (Sending only the MBs that change beyond some threshold).
- Adjusting the INTRA-frame encoding refreshment rate according to the state of the network.
- Requesting INTRA-coded images right after a packet loss has been detected.

This recommendation has also defined two specific-RTCP control messages that are used to speed up the refreshment of the video in case of packet loss, "Full INTRA-frame Request" and "Negative Acknowledgment". These control packets should be used with caution because they can have negative effects if the number of decoders is large.

- Full INTRA-frame Request control packet (FIR) is sent by the decoder whenever a burst of packet loss is detected. The decoder uses FIR packets to refresh the video frame and to speed up the recovery.
- Negative Acknowledgements (NACK) packet is more efficient than the FIR packet because it requests a specific re-initialization of the missing blocks rather the complete refreshment of the image through FIR.

2.2.2 The MPEG Codec

The MPEG [10] standard was developed to meet the growing need for a wider range of applications. MPEG is supported in H.323 systems as an optional codec. MPEG can deliver multimedia at higher rates (2-8 Mbps) and with higher quality. MPEG achieves high compression rate by storing only the changes from one frame to another. It defines three types of picture formats: Intra-Pictures (I-Pictures), Predictive-Pictures (P-Pictures), and Bi-directional Pictures (B- Pictures). It also defines a hierarchy of four layers: Video sequence, Group of Pictures, Picture, and Slice (Group of Macroblocks).

The hierarchical structure of MPEG makes it very sensitive to packet loss. A loss that occurs to one frame can propagate to all the dependent frames and hence disrupts the perceived QoS for a considerable amount of streams [11]. In [8], It was reported that 3% of Network packet loss with MPEG can result into a 30% frame error rate. Different solutions have been proposed to alleviate this problem. Table 2 summarizes these solutions.

Solutions	Description
Filtering	Filtering video streams can be preformed through different mechanisms: Frame dropping, low-pass filtering, color-reduction, and re-quantization. Multimedia-aware Filters need to be placed in the nodes that connect to bottlenecks. They drop packets in case of congestion depending on their importance: P-Frames, B-Frames, and I-Frames are dropped in this order.
Error Recovery	Lost data packets can be ignored, retransmitted, or recovered by a Forward Error Correction (FEC) scheme. Some filters ignore lost packets. Cyclic UDP strategy can be used to retransmit the frames with high priority like I-frames to give them a better chance to reach destination.
Adaptation Algorithms	Those algorithms are similar to TCP congestion control algorithms. The application has two thresholds HIGH-LOSS and LOW-LOSS. The frame rate is adjusted according to those values. If the loss rate is less that LOW-LOSS, it is decreased. If it is more that HIGH-LOSS it is increased[2].

Table 2: Suggested Solutions for MPEG loss issues

2.3 Summary of the Different Control Loops

Table 3 summarizes the different mechanisms that have been implemented to support multimedia transmission in Packet Switched Networks. Table 4 summarizes the main controlled packets discussed in this paper. These mechanisms give the multimedia streams robustness against packet loss.

Class	Mechanisms
The Codec level	Separation of media into different streams
	Interleaving
	State information Carried in all the packets to be able to decode in case of loss
	MB level fragmentation (H.261 packetization)
	Special positioning of the slices in MPEG-1/2 in the RTP packetization
	Switch from Inter-frame to Intra-frame coding
	Control the frame output rate by the audio/video coders (Ex: TCP-like control flow in the IVS system)
	Color to monochrome switching
	Re-quantization
	Mixing of both video and audio in one stream
	Error concealment (FEC mechanism)
	Layered video compression (MPEG and H.263)
The Transport level	Filtering
	Frame dropping
	Retransmission of important frames (ex: the I frames in the MPEG system and the base layer frames in the H.263 system)
	NACK mechanism
	Use of NP_ACK and NP_NACK in MPEG-4
	Subscription to different layers
	Transport prioritization of the different layers (Ex: the base layers are prioritized over the enhancement layer in H.263)

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[2] The rate control algorithms used in the INRIA Video Conferencing System makes sure that the output rate stays within the interval [MIN_RATE, MAX_RATE]. The rate is adjusted as follows:
If Congestion [max_rate = max(max_rate/GAIN,MIN_RATE)]
Else if No_congestion [max_rate = min(max_rate +INC , MAX_RATE)]
GAIN and INC are determined by the application.

	Some H.245 Commands and Indications (See Table 1)

Table 3: Summary Table of the different Mechanism

Congestion Control Packets			
	Control Loops	**Cause**	**Example of Control message(s) Used**
A)	Reduction of the encoder's bit rate	Packet loss exceeds an application specific threshold	H.245 Flow Control Command
B)	Stop the transmission of a specific media stream	If the rate needs to be dropped below the lowest allowed rate of a given stream	H.245 Flow Control Command
C)	Return the Adaptive Busy Signal	When congestion is detected at the level of the receiver through packets loss	Q.931 Release Complete Message
D)	Requesting INTRA-frame refreshment	Burst of packets have been lost	RTCP FIR in H.261 streams
E)	Requesting specific re-initialization of missing blocks	Loss of packets when the number of encoders is small (generally less than 10)	RTCP NACK in H.261 streams
F)	TCP-Like Flow Control	In the case of congestion and non-congestion. The encoder rate is adjusted accordingly	Statistics calculated from RTCP Receiver Report packets

Table 4: Multimedia Control Packets

3. Building H.323-aware 3G Wireless Networks

Supporting H.323 Multimedia standard in a wireless environment requires having robust and efficient congestion and admission control mechanisms given the known characteristics of wireless networks: **1.** Low and variable bandwidth **2.** High error rate **3.** Slow and fast fading **4.** Significant delays and variable delays and **5.** Non-congestion related loss. H.323 QoS signaling has been tailored for fixed networks and hence would require adaptation to the specific characteristics of the wireless link.

3.1 Wireless effects on H.323 QoS Control Loops

Although 3G wireless systems have extended the services currently provided by second-generation systems with high data rates, they still share at a high level the same characteristics of wireless networks and their bandwidth remains relatively very small compared to wired networks. Most of the H.323 QoS control loops would be affected negatively by such characteristics especially that most of them exhibit TCP-like behavior (see A, C, and F control packets in Table 4) and it is a well-known problem that TCP congestion control has failed in wireless systems [12]. Most of the control loops are dependent on packet loss. They assume that packet loss is an indication of congestion and they try to adapt accordingly. However this is a wrong assumption in wireless environments since there is a significant amount of loss that is not related to congestion but rather to mobility, fading, high error rates, co-channel interference, and the sudden transient behavior of the wireless link quality.

The major wireless characteristics that affect H.323 control loops are:

1. Large round-trip times and variance in delay or jitter that are due to the slow wireless link and not to transmission problems. The Jitter field in RTCP reports would be affected and would cause a decrease in the sender frame rate. This would degrade the perceived QoS even to the users that are in fixed networks and who are not suffering from any congestion problems.
2. Non-congestion related losses would trigger an undesirable decrease in the frame rate. A significant amount of packet loss in wireless environments is caused by user mobility, handoff, and co-channel interference. TCP-like congestion control mechanisms, Q.931 adaptive busy signal and H.245 flow control will perform badly in such conditions and will degrade the general perceived QoS instead of improving it.
3. Prolonged fading would result in burst of errors and temporary blocking and would increase the delay perceived by the sender. Again this would affect the perceived QoS.

3.2 Adapting H.323 QoS Control Loops to wireless environments

H.323 QoS Control Loops need to adapt to the specific characteristics of wireless network systems. The strategic location where adaptation of those messages should occur is the edge of the 3G wireless network and more specifically the edge of the core network [13], the 3G Serving GPRS support Node (SGSN), since it is a point of bottleneck in 3G wireless networks and since it is the point of interconnection between data packet networks and 3G wireless networks (see Figure 1). The 3G SGSN, therefore, needs to be H.323 aware and needs to have additional capabilities of seamlessly adapting H.323 signaling to varying wireless network conditions. We propose a filter mechanism at the GGSN that is based on both multimedia feedback through RTCP reports and additional link feedback about wireless link conditions. Link feedback is carried using the Link Feedback Protocol (LFP) as shown in Figure 1. The filter consists of a layer that is added on top of the TCP/UDP layer and before the GPRS Tunnel Protocol (GTP), the layer responsible for tunneling and adding routing information. We call this layer the Multimedia Adaptation layer because it adapts multimedia transmission to wireless conditions as well as multimedia transmission. The MAL takes both wireless link parameters and H.323 control packets and makes a decision whether to forward, stop, or modify the control packets (See Figure 2).

Multimedia feedback is not enough to trigger the right codec adaptation. As stated earlier, many H.323 control messages are interpreted wrongly at the level of the receiver. Therefore there is a need for link feedback: (Signal/Interference ratio, Cost, Power, flow queues, handoff, etc.). The filters are capable of understanding both link and multimedia feedback and can act intelligently at both the codec and the transport level (see Figure 3). Similar to multimedia feedback, link feedback should be set periodically to keep the 3G SGSN filter aware of the current network conditions. Link feedback is necessary to help filters predict wireless link conditions and act accordingly on the control loops. This would help the filter distinguish between congestion-related and non-congestion related loss. Table 5 summarizes the actions that need to be taken in the case of non-congestion related loss.

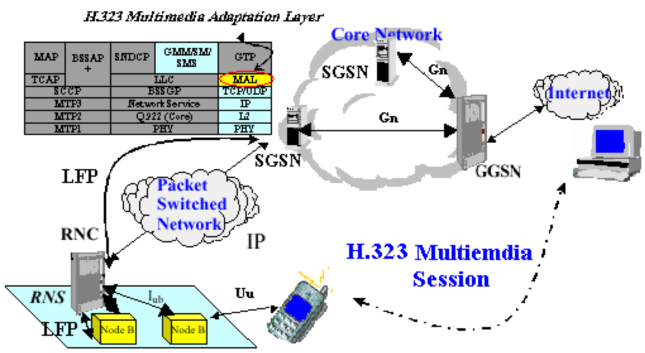

Figure 1: Filter Location in a 3G wireless Network

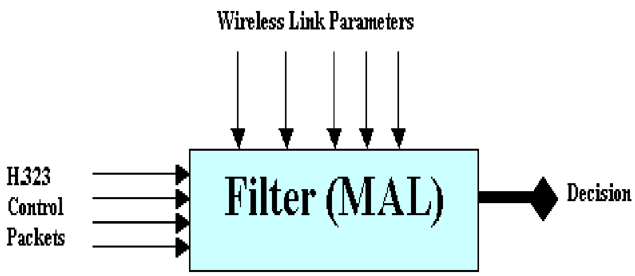

Figure2: Filter Input/Output

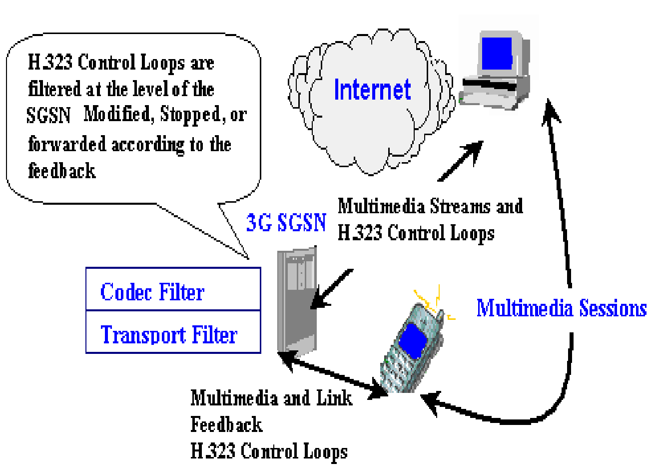

Figure 3: Filter Architecture

H.323 Control Packets	
Control Loops	**Filter Actions in the case of non-congestion**
Reduction of the encoder's bit rate	Stop
Stop the transmission of a specific media stream	Stop
Return the Adaptive Busy Signal	Stop
Requesting INTRA-frame refreshment	Forward
Requesting specific re-initialization of missing blocks	Forward
TCP-Like Flow Control	Stop/modify

Table 5: Filter Actions on H.323 Control Loops in case of non-congestion

4. Conclusion and Future Work

In this paper, we have discussed the QoS issues that are involved in H.323 networked multimedia applications. Some of the main control loops have been identified and discussed. The problems that might occur with networked multimedia systems have been put in context. We have proposed a solution that consists of building H.323 Filters at the level of the Serving GPRS Support Node (SGSN) that act intelligently on the different control loops depending on both link and multimedia feedback from the wireless end stations. We are currently conducting experiments to detect the identified control loops in a third generation wireless environment [2].

This work lays down the floor to future investigations. The control loops identified in H.323 multimedia networks need to be accounted for in future Base Stations. The filter architecture needs to be implemented for evaluation and testing purposes. This would provide us with a good framework for developing H.323 aware multimedia 3G wireless systems.

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