SUBGROUP FEEDBACK FOR SOURCE-SPECIFIC MULTICAST

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Abstract: The recent deployment of IP-based TV and radio distributions requires one-to-many multicast instead of the traditional many-to-many data distribution. These large multimedia sessions usually rely on the real time protocol (RTP) and the real time control protocol (RTCP). Although one-to-many multicast offers the required communication, it does not support the multicast feedback channel for carrying the RTCP control messages. Therefore, unicast feedback channels from session members to the source are used to carry these messages. In this paper, we introduce subgroup feedback scenarios for source-specific multicast, which is built on the one-to-many philosophy. Our extensions are based on the subgroup feedback framework standardized in the IETF. We outline a possible implementation of the subgroup feedback using the receiver summary information (RSI) packet. A theoretical RSI packet rate analysis is also presented in the paper.

1 INTRODUCTION

New IP-based broadcasting services currently being deployed do not require many-to-many communication offered by the well-known any source multicast (ASM) (Quinn and Almeroth, 2001). As a result, instead of many-to-many, a new one-to-many philosophy is preferred. To support this, source-specific multicast (SSM) (Holbrook and Cain, 2004), (Bhattacharyya, 2003) has been developed. SSM is expected to cover all types of IP-based multimedia sessions with many receivers and only one source, such as IPTV broadcasting. Multicast sessions (either ASM or SSM) typically rely on the RTP/RTCP (Real Time Protocol/ Real Time Control Protocol) protocol (Schulzrinne et al., 2003). The RTP flow carries multimedia data, whereas the RTCP flow carries signalization and synchronization. RTCP, which is more sophisticated than RTP, provides a set of messages exchanged among session members. These messages are used as a feedback for monitoring the session behavior. The session feedback could be used, for example, for the parameterization of multicast forward error correction (FEC) or the tuning of suppression algorithms. The feedback flow usually involves information such as a summary of data sent, synchronization of different transmitted media (audio, video), packet loss, packet delays, and interarrival jitter. This information can be also used, for example, to localize distribution problems for particular receivers. Furthermore, a third-party application could receive the feedback flow for long-term session monitoring.

However, SSM lacks the support for communication among session members (i.e. many-to-many). Therefore, RTCP packets cannot be transmitted since SSM does not offer the required feedback channel. Feedback RTCP data are transmitted using several packet types. The two basic types are sender report SR (it carries transmission and reception statistics) transmitted from the source to receivers, and receiver report RR (it carries reception statistics) transmitted from receivers to the source. The existing solutions use unicast feedback connections from receivers to the source. The two known methods are: reflection and summarization (Chesterfield and Schooler, 2003). Both methods deal with the same transmission of RR packets from receivers back to the source, but they differ in processing the packets received at the source. After processing the received data, the feedback data are sent back via SSM to all session receivers, as depicted in Figure 1. With the reflection method, the source forwards every RR packet to all session receivers. However, the forwarding could be harmful to the network load, especially with the session size growing. In addition, the source does not need to forward all the received data – some of them are necessary only for the

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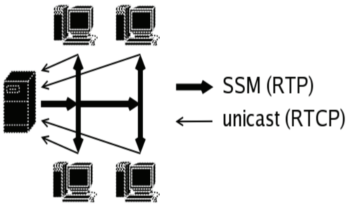

Figure 1: Unicast feedback with SSM.

source, not for the receivers. On the other hand, problematic receivers can be easily identified and certain actions can be taken to improve the session quality. The reflection method is backward compatible with currently used RTP/RTCP implementations. The traffic generated by the source, when RR packets are forwarded, is not regarded as traffic belonging to the source.

The summarization method deals with an aggregation of selected receivers' data from RR packets in the source. When an aggregation is finished, a summary packet is assembled and sent to all receivers via multicast. Several known algorithms can be used for aggregating the received values (Chesterfield et al., 2004). In addition, the aggregated values can be compressed up to a factor of 16. The compression significance is growing when large sessions are being used. The resulting summary information is conveyed using the Receiver Summary Information packet (RSI) (Chesterfield et al., 2004) containing several elementary fields for encapsulating the carried information, such as timestamps and group size. The RSI packet also has several optional sub-report blocks that follow the RSI packet: loss, jitter, cumulative loss, general statistics and receiver bandwidth. These sub-report blocks are considered an enhancement to the key information required by receivers. The source sends one RSI packet per one reporting interval. The RSI packets are sent together with the sender SR packets.

2 FEEDBACK RATE PROBLEM

One of the major problems with RTP/RTCP is the large delays between sending RTCP packets from a session member (Rosenberg and Schulzrinne, 1998). As mentioned above, RTCP provides feedback about the multimedia session quality. This information is usually used by high-layer protocols to control and monitor the session behavior. Thus, the session control is strongly affected by the feedback reporting frequency. This means that even minimally outdated values can burden further data processing with an

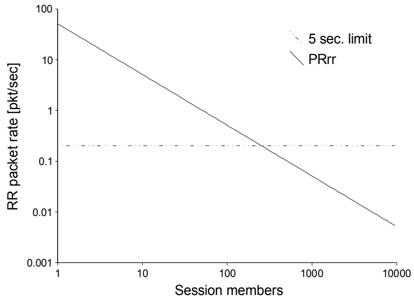

Figure 2: RR packet rate vs. session members for 1Mbps session bandwidth.

error and consequently the behavior of the session can be faulty. Therefore, the RTCP packets (either SR or RR) should be sent as often as possible. On the other hand, many RTCP packets can load the assigned session channel too much. Besides that, there is a possibility that not enough bandwidth is left for the multimedia flow carried by RTP packets. In case of large sessions, RTCP packets can interfere with RTP packets and cause packet delay and loss. Therefore, a mechanism to control the frequency of RTCP packets is used. The algorithm used keeps the frequency of RTCP packets at a value corresponding to 5% of the total allowed session bandwidth. Furthermore, 3.75% of the session bandwidth is used by RR packets and 1.25% of the session bandwidth is used by SR packets.

With an SSM session, the RR packet rate PR_{rr} is thus calculated as

$$PR_{rr} = \frac{0.75 \times BW_{rtcp}}{PS_{rr} \times n_r} \quad (1)$$

and the SR packet rate PR_{sr} is identified as

$$PR_{sr} = \frac{0.25 \times BW_{rtcp}}{PS_{sr} \times n_s} \quad (2)$$

where PS_{rr} is the average size of the RR packet, PS_{sr} is the average size of the SR packet, BW_{rtcp} is the allowed bandwidth used for RTCP packets, n_r is the number of receivers, and n_s is the number of senders of an SSM session. The formulas assume that the number of senders is less than or equal to 25% of session members - an SSM session has only one sender. Also, it is assumed that the session has four receivers or more. The algorithm used gives 25% of the RTCP bandwidth to senders and 75% of the RTCP bandwidth to receivers.

For the purpose of feedback rate problem description, it is necessary to determine the RR packet size. Using the values presented in

(Chesterfield and Schooler, 2002), the average RR packet size is set to 736bits. As an example, the identified RR packet rate for 1Mbps session bandwidth is depicted in Figure 2. The graph reveals that with a great number of receivers the RR packet rate is very low. For example, for 10 000 session members the rate is 0.005 pkt/sec (200 seconds per member). However, this number of members is expected when certain services are provided, such as IPTV broadcasting. Thus, some values reported by session members could lose their importance if they become too old. This problem has been identified as one of three major problems with RTP/RTCP - a low rate of session member reports when the session grows extremely large. Furthermore, the dashed line in the graph shows the minimum transmission interval between RR packets – 0.2pks/sec (5 seconds per member). The purpose of the limitation is to avoid packet floods when the session behaves unexpectedly (Schulzrinne et al., 2003). For instance, when a session is experiencing a network part failure, the number of session members could become extremely small during a short period of time.

With the RTP/RTCP multicast, not all the feedback data from session members are of equal importance. This means that some member feedback needs to be reported more frequently. For that purpose, a biasing algorithm was developed for SSM with the summarization method (Chesterfield and Schooler, 2003). The algorithm is built on sampling the feedback for a specific range of values. Then, members reporting values within this range are treated as biased. The remaining members are unbiased. The source conveys to the receivers its selection range. For the biased group of members, for example, the source can increase the amount of allowed bandwidth and decrease the bandwidth for the unbiased members.

3 SUBGROUP FEEDBACK SCENARIOS

In this section, we propose two new subgroup feedback scenarios for SSM sessions with the sender summary model. For simplicity, we assume one subgroup within a session. Our proposal is based on two possible uses of the multicast subgroup feedback – the hierarchical aggregation used to reduce SSM feedback latency (Chesterfield and Schooler, 2003) and multimedia transmission monitoring application. With these applications, the feedback from the subgroup members is more important to receive than any other feedback. In the case of using the hierarchical aggregation, a subgroup member acts as summarization server for a group of other multicast members. Members from this group report theirs feedback to the summarization server and the server puts the feedback into a single summary. Summarization servers are members of another group placed higher up. As other multicast members join or leave the session, the number of remaining members in this group varies. In the case of using a transmission monitoring application, the subgroup members could be, for example, constant sensors reporting rough values of the whole monitored multimedia system. As other sensors join or leave the session, the number of remaining session members again varies. Figure 3 shows possible scenarios for the subgroup feedback framework. The three following cases are considered:

- **Case a)** A subgroup sends RR reports to the IP feedback address of the source

This scenario was previously described in the biasing method. The source receives the feedback values from both the subgroup and the remaining session members. The valuable feature is that the feedback rate of subgroup members could be adjusted as desired. However, the 5% limit of the session assigned bandwidth has to be achieved. With this scenario, for example, the source can specify a subgroup with problematic session members and speed up their feedback rate or slow down the feedback of other session members whose feedback is not so important. A monitor application shown in the figure receives the feedback data from all session members via multicast. Therefore, the reported values cannot be assigned to a particular session member.

- **Case b)** A subgroup sends RR reports to the IP feedback address of a LAN monitor

Now the sender does not receive the subgroup feedback. Since the feedback rate from the subgroup is outside the session bandwidth control, it is possible to speed up the feedback rate as desired, outside of the 5% restriction. This scenario is expected to be applied in a multimedia transmission monitor application. Also, this scenario brings an advantage of comparing the feedback data from all session members with data received only from the subgroup. In this way, a subgroup sample could be

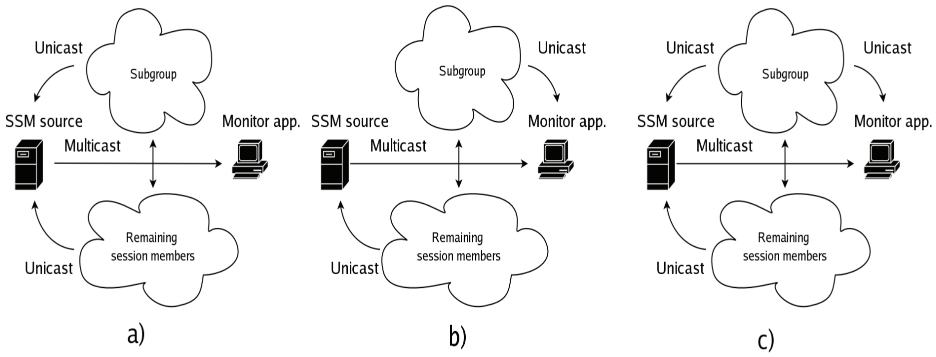

Figure 3: Subgroups feedback scenarios.

set up and used for describing the whole session behavior. The subgroup size should be set carefully with certain applications. A large subgroup could make the whole session behavior faulty since not enough feedback is provided for the source. A monitor application could receive feedback values from the whole session and also from the subgroup members.

- **Case c)** A subgroup sends RR reports to both a LAN monitor and the source

The third case deals with the reflection of the subgroup feedback data from session members. The session feedback acts as described in case a) and furthermore a new feedback rate above the 5% limit could be defined for a LAN monitor.

For the purpose of specifying a subgroup, the feedback reported values could be used. Feedback data for RTP/RTCP sessions are introduced in (Schulzrinne et al., 2003). They are, among other things, loss distribution, jitter distribution, round trip time distribution, cumulative loss, and general statistics. Furthermore, we use an additional subgroup parameter, the feedback sender IP address, for a session member localization. These subgroup parameters could be mixed to create a subgroup defined by several requirements. As with the biasing method, subgroup rules based on the selected feedback parameters are defined. The source transfers the rules to session members. When a session member receives the defined rules, it compares them with own feedback reported values. If the feedback data meet the subgroup required criteria, a member joins the subgroup.

3.1 Implementation of Extended Subgroup Feedback

The source informs session members that its reporting values are or are not within the desired range to join a subgroup, as introduced above. Since a session member cannot be identified in the sender summary report model, it is not possible to send such information to a particular member. Therefore, we have adopted the RSI packet used purely in the sender summary report model. The RSI packet consists of various types of information encapsulated in optional sub-report blocks that follow after the fixed header. Optional sub-report blocks use a generic format that includes sub-report block size (SRBT), length and sub-report specific data. Furthermore, to carry distributions of values (i.e. loss, RTT), a data bucket format is used. This format divides data into variable length buckets. Among other things, minimum and maximum distribution values are included to indicate the data bucket range. We have extended the RSI packet in order to allow subgroup parameters to be encapsulated in a new RSI packet sub-report block called session subgroups. This new packet block consists of the following fields: SRBT, length, number of subgroups, and subgroup description. A subgroup description format is: length, number of members, IP feedback address, IP feedback port, IP feedback address flag, and allowed bandwidth. The feedback flag says whether the feedback traffic is reflected to both the source and the feedback address or only the feedback address is used. What follows is the subgroup rules formatted as number of rules and rule description. The rule description consists of the ID number of the parameter. We use IDs as defined in (Chesterfield et al., 2004) and parameter minimum and maximum values for the rule parameter. The number of rules per group is limited to 255. The packet structure is depicted in Figure 4.

3.2 Analysis of the Adopted RSI Packet Rate

The sender should send at least one RSI packet for each calculated reporting interval, as was given in (Chesterfield et al., 2004). However, the RSI packet sub-report blocks are optional as we mentioned above. It is recommended that no more than 20% of the bandwidth assigned to a session member is used to carry the optional information in RTP/RTCP sessions. Packet rates above this limit could result in protocol misbehavior. Therefore, the adopted RSI packet rate should be lower than the original one. To meet this requirement, we have recalculated the adopted RSI packet rate. Considering that there is only one sender in an SSM session, the packet rate PR_{sr} for sender reports is from equation (2) identified as

$$PR_{sr} = \frac{0.25 \times BW_{rtcp}}{PS_{sr}} \tag{3}$$

provided that there are at least four session members. Supposing that PR_{rsi} is equal to PR_{sr} and 10% of the bandwidth assigned to the sender is used to carry optional information, the packet rate for the adopted RSI packet PR_{rsi_adp} can be calculated as

$$PR_{rsi\_adp} = \frac{0.025 \times BW_{rtcp}}{PS_{rsi\_adp}} \tag{4}$$

where PS_{rsi_adp} is the size of the adopted RSI packet. Based on equation (4), the adopted RSI packet size must be included in the calculation. Thus, we assume these field lengths of the adopted RSI packet (Chesterfield et al., 2004): fixed header - 160bits, SRBT - 8bits, length - 8bits, number of subgroups - 8bits. A subgroup consists of length - 32bits, subgroup members - 32bits, IPv4 feedback address – 32bits, IPv4 feedback port – 16bits, feedback flag – 1bit, allowed bandwidth – 32bits and the number of rules - 8bits. A subgroup rule consists of the ID number - 8bits, minimum value – 32bits and maximum value - 32bits. Therefore, for the purpose of theoretical calculations, the estimated average packet size PS_{rsi_adp} is

$$PS_{rsi\_adp} = 184 + n_{sgr} \times (145 + n_{rul} \times 72) \tag{5}$$

where n_{sgr} is the number of subgroups in a session and n_{rul} is the number of subgroup rules. Figure 5 examines the calculated adopted RSI packet rate with $n_{rul} = 3$ and session bandwidth = 64Kbit/s.

The dashed line in the graph shows a minimum transmission interval between adopted RSI packets – 0.2 pkt/sec (5 seconds). The purpose of the

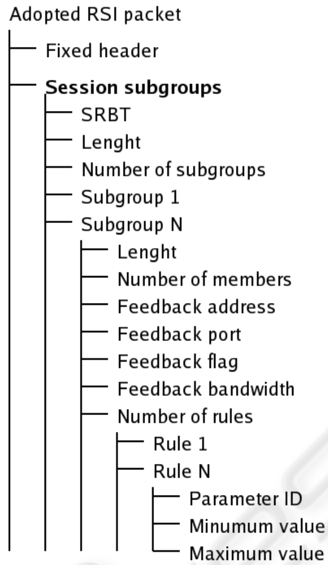

Figure 4: Adopted RSI packet.

limitation is to avoid packet floods when the session behaves unexpectedly, see (Schulzrinne et al., 2003). For instance, when a session is experiencing a network part failure, the number of session members could be extremely small during a short period of time. When the values go over the limit, equation (4) has to be redefined as

$$PR_{rsi\_adp} = min\left(0.2, \frac{0.025 \times BW_{rtcp}}{PS_{rsi\_adp}}\right) \tag{6}$$

4 CREATING A NEW SUBGROUP

For the purpose of creating a new subgroup within an SSM session, we have chosen the application-defined (APP) RTCP packet introduced in (Schulzrinne et al., 2003). The packet is intended for experimental use. In our case, the packet carries parameters of a subgroup to be created from a session member to the source. The field called application-dependent data is used for this purpose. The data format is the same as we introduced in section 3.1. When a session member sends a new subgroup request, the source accepts or denies the creation of the subgroup. The source should allow only certain session members to create a new subgroup in order to avoid subgroup overflow. Furthermore, creating duplicate subgroups has to be considered. Therefore, requests for creating a new subgroup are acknowledged by the source. For the acknowledgment, we again use the APP packet. If the session member does not receive the

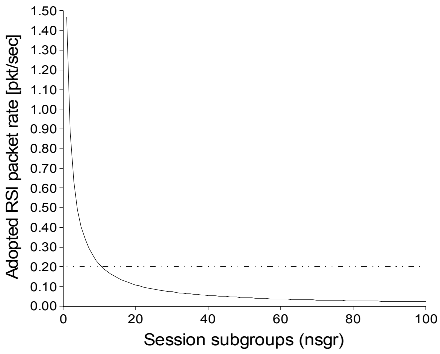

Figure 5: Adopted RSI packet rate.

acknowledgment, the request is sent again within a selected period.

Also, we assume that an application itself is responsible for the authorization of allowed members to create a new subgroup. Besides this, there are quite different requirements on the subgroup creation process. For example, with a monitoring application, only one subgroup is assumed. On the other hand, with the hierarchical aggregation, the number of subgroups depends on the number of aggregations levels.

5 CONCLUSION AND RELATED WORK

Our work has been inspired by one of three major problems with RTP/RTCP that deals with the low rate of the session member reports when session grows extremely large. In this paper, we extended the subgroup feedback framework for SSM based on the biasing algorithm for the sender summary model. The two new subgroup feedback scenarios allow subgroup members to change their feedback rate in several ways as desired. A possible use of these scenarios is the hierarchical aggregation for SSM (a subgroup member atcs as summarization server for a group of multicast ordinary members) and multimedia transmission monitoring application that uses the multicast feedback channel to report monitored values (the subgroup members are constant sensors reporting rough values of the whole monitored system). A new feature is the comparison of subgroup feedback data with feedback from all the session members. For the purpose of transferring the subgroup parameters from the source to session members, we modified the RSI packet by adding new specific fields. Since the new fields in the adopted RSI packet are optional, we defined a packet rate that meets the requirements for optional data transmission used in RTP/RTCP sessions. For the purpose of describing the adopted RSI packet rate, a theoretical analysis was also presented in the paper.

Our related work deals with setting the appropriate thresholds for subgroup parameters. Some feedback values change quite often within a wide range, for example, packet loss, delay and jitter. Subsequently, a session member changes its subgroup identity as these feedback values are above or below the defined threshold. Thus, session members should take care when processing the subgroup parameters. We also aim to extend the subgroup identification with the parameters introduced in (Friedman et al., 2003) and (Ott et al., 2004), such as jitter buffer parameters, configuration parameters, bit vector chunks, and slice loss indication.

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