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Network-driven Adaptive Video Streaming in Wireless Environments

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Abstract—**In this paper, a concept of a network-driven adaptive video streaming is proposed and investigated in a state-of-the art UMTS/HSDPA network in order to alleviate the impact of strongly time-varying quality of wireless links on video quality. We propose making adaptation decisions in a radio access network based on channel-quality information available there. This is done by a so-called Multimedia Network Controller (MNC), which is aware of server and client capabilities and the channel quality. Upon a decision, the MNC sends a rate switching request to the video server, which adapts its data rate. We evaluate results with a novel PSNR/video-interruptions diagram allowing to show and to compare video quality graphically in a compact form. For the performance evaluation, we developed a comprehensive video streaming testbed around a detailed UMTS/HSDPA emulation tool.**

I. Introduction

During the recent past, mobile cellular telecommunication networks have been developing strongly and continuously. New emerging technologies and generations of mobile networks, such as the recent High Speed Downlink Packet Access (HSDPA) for UMTS offer higher data rates in radio access networks in comparison to previous technologies. Caused by these new technical developments, new services are offered in mobile networks, which have only been available in fixed networks until recently due to bandwidth limitations. One such service is the non-interactive video streaming.

In general, video streaming applications in a mobile wireless environment encounter highly time-varying and error-prone radio channels. In spite of measures taken on the physical and the MAC-layer , such as ARQ mechanisms and scheduling, the disturbing impact of the wireless transmission may eventually lead to a degradation of the perceived quality for a video service with real-time requirements. Therefore, additional efforts are needed on the network, transport or application layer.

Generally, measures targeted on improving video streaming QoS can be grouped into four major approaches: resource reservation, traffic prioritization, optimized resource usage and application-layer QoS control, whose one representative, content adaptation, is in focus of this work. Whereas the first three approaches deal with the usage of network resources, the content adaptation is aimed at decreasing transmission errors by reduction of the video-data volume to be transferred.

There exist two fundamental techniques of the video-content adaptation to the time-varying transmission resources: Dynamic Rate Shaping (DRS) and Rate Control (RC). DRS performs the content adaptation of a video content without an interaction with a video server [1]. A shaper is a ”filter” producing an output video stream by changing the input stream according to the dynamic data rate constraints. One example of it is application-aware queue management in the network [2]. In contrast to DRS, the aim of rate control is to make the content server/encoder change the video data rate according to the current available resources [3].

The standard RC implementation is an RTP/RTCP-based end-to-end solution. Its problem is imprecise and slow implicit measurement algorithms of fast-changing channel quality, which results in wrong adaptation decisions. In this paper, we propose making adaptation decisions in a Radio Access Network (RAN) based on channel-quality information available there. This is done by a so-called Multimedia Network Controller (MNC), which is aware of server and client capabilities and the channel quality. Upon a decision, the MNC sends a rate switching request to the video server, which adapts its data rate. We evaluate results with a novel PSNR/video-interruptions diagram allowing to show and compare video quality graphically in a compact form. For the performance evaluation, we developed a comprehensive video streaming testbed around a detailed UMTS/HSDPA emulation tool.

This paper is organized as follows: Section II gives an overview of related work in the area of the network-supported rate control. In Section III-A, we introduce the idea and advantages of the MNC. The considered system and scenario are presented in Section IV. We describe the metrics used for the video quality evaluation in Section V. The performance of the network-driven adaptive video streaming is evaluated in Section VI. Section VII concludes the paper.

II. Overview of Network-supported Rate Control

To improve the adaptation ability, some few architectures supporting direct information exchange between a network and a server were proposed in the past.

In [5], this idea was mentioned and referred to as *network congestion feedback*. In [6], a concept of a so-called *streaming agent* (SA) located at the junction of a wired network and a wireless link was developed. This streaming agent sends timely feedbacks to the server indicating whether every packet has arrived at the SA correctly in time. Therefore, the SA can quickly notify the server about congestion in the *wired* network. Simultaneously, RTCP feedback from the client (connecting to the wireless link) is also used. The server combines the information from both reports and makes its adaptation decision. In [7], a so-called *network-information-based* RC is presented, where end-to-end RTCP-receiver reports and intermediate network information are combined by the video server in order to make an adaptation decision. The network information is obtained from intermediate nodes by means of Simple Network Management Protocol (SNMP) and Management Information Base (MIB). In [8], an idea of *Radio Network Feedback* (RNF) in a UMTS network is described. The concept of RNF is based on immediate and explicit reports to the video server about the current wireless link condition in a UMTS Teresstrial Radio Access Network (UTRAN). It was shown that the RNF-based adaptation is faster and more precise than the end-to-end solution.

In contrast to these approaches, we propose making the adaptation decisions by the owner of resources, i.e. the RAN, taking server and client capabilities into account. This idea is detailed in the next section.

III. Multimedia Network Controller

A. General Idea

The idea of a Multimedia Network Controller (MNC) proposed here is to control video streaming QoS from a RAN (Figure 1). The MNC is *aware* of video-client and server capabilities. Generic tasks of the MNC are: observation, estimation, decision/measures and control communication, which is detailed in the following.

Observation: the wireless link quality and the wired part of a multimedia connection are permanently observed. For the wireless part, the following parameter vector may be observed: instantaneous values of the link-layer data rate, the number of users in a cell, the available radio power, the radio-link buffer occupancy. The observation of the wired part can be done by RTP/RTCP.

Estimation: Due to high time-variance, instantaneous values of the parameters observed cannot be used for making any decisions. Estimation algorithms are necessary in order to recognize main trends for the short-term future.

Decision and measures: The MNC makes its decision according to client and server capabilities. Examples for the client capabilities are maximum decodable video-data rate, display resolution and so on. A server can be capable of adapting or not and so forth. On the other hand, the global sight of the cell can be used for optimization tasks. Examples of possible measures to be taken may be rate control, rate shaping, cell optimization.

Control communication: It serves for exchange of capabilities information, observation of the wired part of the connection and notifications to the end systems.

To sum it up, the advantages of the MNC are the access to reliable information about channel quality and the global sight of the cell, which allows to choose and take proper measures according to the operating point of the RAN. The awareness of the client and server capabilities allows to make the best possible decision.

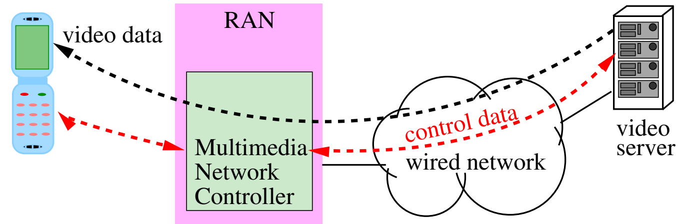

Fig. 1. basic principle of MNC.

B. Realization of MNC for network-driven adaptive Streaming

According to the above general MNC tasks, the following MNC model (Figure 2) was implemented in the Node B and in the Radio Network Controller (RNC) of the UTRAN model described in Section IV.

Observation: HSDPA provides adaptive modulation and coding as a measure to deal with time-varying channel quality. This results in permanently adjusting link data rates.

The MNC observes separately every video streaming connection by polling gross link data rates for every radio-link scheduling interval in the MAC-hs layer (r_s). Due to the shared nature of the HSDPA link, the number of users in the cell (n_u) and the available radio power (as power factor k_{pow} that is a function of n_u) are also considered for the determination of the gross instantaneous link data rate of one user (r_u):

$$r_u = k_{pow} r_s$$

Due to power settings of our Node B, k_{pow} was determined as follows:

$$k_{pow} = \begin{cases} 1 & : \quad n_u \leq 2 \\ 2/n_u & : \quad n_u > 2 \end{cases}$$

Our model assumes that the wired part of a video connection has enough resources. Therefore, it is not necessary to observe it.

Estimation: An estimation algorithm calculates a single value (estimation value e_i) with the help of a series of measurement values (m_i). For a bandwidth estimation, the

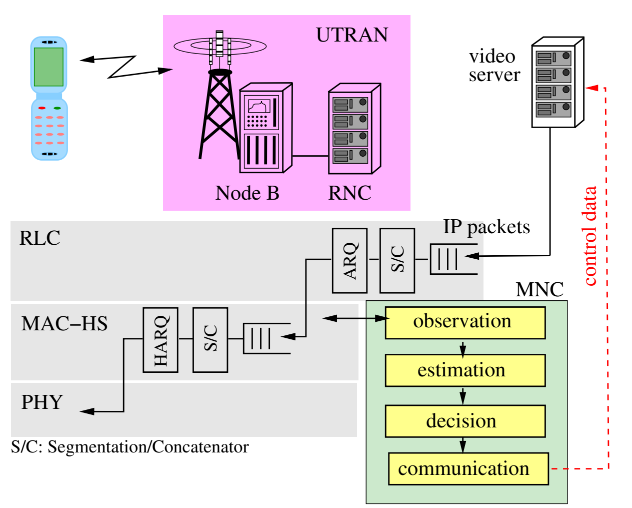

Fig. 2. MNC implementation in UMTS/HSDPA RAN.

exponential moving average can be used with satisfying results [13]:

$$e_i = (1 - A)e_{i-1} + Am_i$$

It is not easy to find a proper value for the parameter A. In our work, $A = 0.04$ was empirically (and with the help of [14] and [15]) identified as a good choice.

Decision and measures: In this work, a Rate Switching Request (RSR) from the UTRAN to the video server is proposed. The decision about the rate switching is based on the following data and constraints:

- estimated radio link rate (r_e).
- available video bit rates on the server side (vr_{avail_i}).
- current video bit rate $vr_{current}$.
- decoding capabilities of the client. (maximum decodable data rate, r_{maxDec}).
- frequently alternate up-down switching can be annoying. To meet this constraint, a hysteresis factor ($k_{hys} = 0.1$) was used as a parameter of the rate-switching decision.

We switched up to vr_{avail_i}, if $vr_{current} < vr_{avail_i}$ *and* $r_e \geq (1+k_{hys})vr_{avail_i}$ *and* $vr_{avail_i} \leq r_{maxDec}$. We switched down to next lower vr_{avail_i}, if $r_e \leq (1 - k_{hys})vr_{current}$.

Due to scheduling of mobile users in the UTRAN, TCP-friendliness does not need to be considered.

Control communication: The MNC sends the RSRs as a plain UDP payload. At the beginning of a video streaming session, the server and the client notify their capabilities to the MNC.

IV. System Model

A. System overview

We consider a single-cell HSDPA-environment, where several User Equipments (UEs) connect to the Node B via a High Speed Downlink Shared Channel (HS-DSCH) in the downlink and a dedicated channel (DCH) in the uplink direction. The Node B is connected to the RNC, which itself is connected to the Internet via the 3G-SGSN and 3G-GGSN of the cellular system's core network. The UEs establish a video streaming connection with a host in the Internet. The Internet and core network were assumed to introduce a constant delay $T_{\text{INet}} = 20\,\text{ms}$ in each direction and not lose any IP-packets.

B. Emulation Model

The HSDPA network was modeled with all its relevant RLC, MAC-d and MAC-hs protocols. The physical layer was modeled based on BLER-curves (BLER stands for BLock Error Rate) obtained from physical layer simulations including HARQ (Hybrid ARQ). Transport formats on the MAC-hs layer were selected based on the channel quality such that the BLER is 10%. We assumed ideal conditions for the reporting of channel quality indicators from the mobile terminals to the Node B, i.e. zero delay, in order to avoid side effects. The Iub flow control was operating with a short update period and no deadtime, since higher values can introduce unpredictable delays and delay spikes [9]. The maximum number of MAC-hs retransmissions was limited to $R_{\text{max,hs}} = 4$, and the RLC layer was operated in unacknowledged mode. We neglect the convergence layer, as it only introduces a very small overhead in a single-cell environment. For a more detailed description of the model, please refer to [10].

The model was implemented with the event-driven simulation library IKR SimLib [11]. This model was then extended as an *emulation* based on the IKR EmuLib [12].

C. Video Server and Video Client

For these investigations, a software was developed that streams an MPEG-4 video content over UDP through the UMTS/HSDPA-emulator. Upon RSR arrivals, the server performs the rate switching. For this, bit stream switching in form of multiple pre-encoded file switching [4] is used. Different bit rates were achieved by a variation of data volume per pixel (i.e. all video versions have the same video frame rate and spatial resolution). Switching occurs only at I-frames [16], which appear periodically.

On the client side, an IP-packet capturing software receives the packets. The received video is reconstructed and evaluated offline. The evaluation of the video quality is carried out frame-based: if any packet belonging to a frame is lost, the whole frame is considered as lost. Further, all frames depending on a lost frame, are also considered as lost. This approach is widely used [17][18][19][20]. The computation of the video data loss due to too late frame arrivals at the client is performed with a cumulative jitter concept. [21].

D. Scenario

The scenario used for our performance evaluation comprises two video clients in a single UMTS/HSDPA cell. The play-out buffer was set to 6 s. The channel is modeled by a realistic urban channel trace (Stuttgart), whose mean gross data rate is $1055\,\text{kbps}$. The video server streams a single-layer MPEG-4 video encoded with the free available ffmpeg software [25]. The video material is a multiple rerun of a short CIF-sequence of an American football match [26]. The transmitted video consists of 33120 video frames. It was encoded for two average data bit rates: 308 and $900\,\text{kbps}$. The video frame rate is 25 fps. The emulation duration is equal to the video length (Figure 3).

Terminal mobility was modeled taking into account both slow and fast fading. All mobile terminals move at a speed of $v = 30\,\text{km/h}$. They periodically experience the same slow fading profile, where each mobile starts at a different position of the profile in order to obtain independent channel conditions in-between the mobiles. A proportional fair [27] scheduler was used at the MAC-hs layer to assign resources to the different data streams.

V. Video Quality Evaluation

The quality of a video is highly subjective. In order to evaluate the perceived quality of a video, a large number of test persons is needed, who are asked to judge the video by using a given score range. One such score range is the well-known Mean Opinion Score (MOS), which is defined by ITU-T in

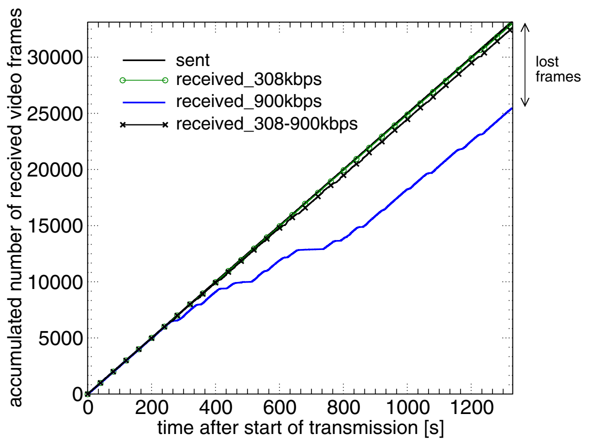

Fig. 3. accumulated number of received frames.

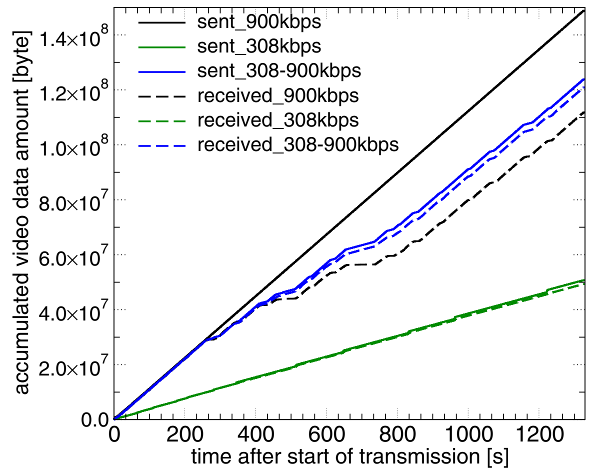

Fig. 4. accumulated video data amount.

[22]. The result of a test viewing is a *subjective* video quality statement. However, this procedure is very costly.

As an alternative, it is desirable to have an *objective* video quality metric that can be obtained directly from the received video data without the need for the test persons. Ideally, this objective metric can then be correlated with a subjective quality metric. A common objective metric is the Peak Signal to Noise Ratio (PSNR), which is also standardized by ITU-T in [23]. Table I shows the mapping of PSNR to MOS [19].

TABLE I
POSSIBLE PSNR-TO-MOS MAPPING

PSNR [dB]	MOS
> 37	5 (Excellent)
31 - 37	4 (Good)
25 - 31	3 (Fair)
20 - 25	2 (Poor)
< 20	1 (Bad)

PSNR is widely used to determine the quality loss due to an MPEG encoding process. The distortion pattern of it is different from the distortions that occur due to packet loss in a network, which rather influences video continuity.

In our investigations, the error concealment technique used at the client is the replacement of lost frames by the last correctly received frame (frame-freezing). To evaluate the video continuity, we used the video interruption length (length of frame-freezing) as an additional metric. According to [24], the interruption lengths above 1 s are unacceptable in any case. To keep the evaluation simple, we want to avoid a finer breakdown of the interruption lengths into a subjective score range.

We used characteristic values of statistic distributions of the above metrics (PSNR and interruption length) - average, lower quartile (Q_1), median (Q_2) and upper quartile (Q_3) to be precise. Below Q_1 lies the lowest 25% of the data; Above Q_3 are the top 25%. Median is the middle of the distribution (50% of the data are above it and below it respectively).

We combined PSNR and the video-interruption-length metric in a novel PSNR/video-interruptions (PVI) diagram allowing to show und compare video quality graphically in a compact form (Figure 5 and 6). The diagram is divided into several areas according to the described score range of PSNR and video interruption lengths. The detailed explanation of this diagram will be given in Section VI-B.

VI. Performance Evaluation

In this section, we compare network-driven adaptive video streaming session with non-adaptive sessions. The disadvantages of RTP/RTCP end-to-end solution was shown in [8].

A. Throughput at the Application Layer

Figure 3 shows the accumulated number of video frames *completely* received by the client during the video streaming session over the time. The deviation from the gradient of the reference curve "sent" indicates the amount of frame loss. The time-shift of the curves to the reference due to network delay is not visible because of a coarse resolution of the time axis. The difference between the end point of a certain curve and the end point of the reference corresponds to the total video frame loss.

We see that the non-adaptive session with 900 kbps lost approx. 27% of its frames. Its rate lies tightly under the average channel rate. This emphasizes the high time variance of the channel. 85% of the lost frames were directly lost in the network (it accords to approx. 23%). The residual loss is the loss due to frame dependencies (MPEG loss propagation) and dead-line exceedances. The IP-packet loss in the network was 15%. Due to inter-packet dependencies, the frame loss was higher (23% mentioned above).

The adaptive session with two rates (308 and 900 kbps), which was controlled by the proposed MNC, results in a good video continuity. The total frame loss is approx. 3%. The RSRs were sent quickly and were precise enough in order to adapt to changes of the radio link conditions.

Figure 4 demonstrates the accumulated video data volume *correctly* received by the client at the application layer. Generally, the amount of video data is a good indicator for the video quality (while using the same encoder). The difference

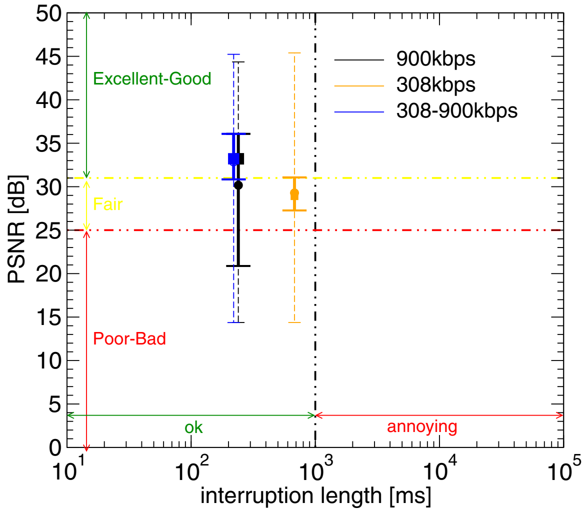

Fig. 5. PVI diagram (only PSNR values are plotted).

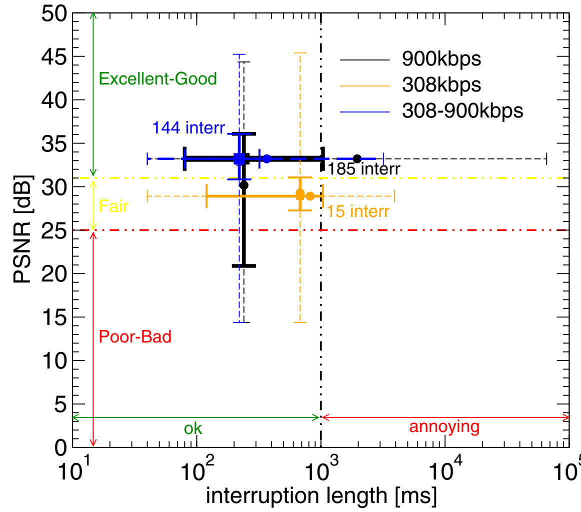

Fig. 6. complete PVI diagram.

between end points of accordant "sent" and "received" curves indicates the amount of lost data. Both curves of the adaptive session lies closely to each other, whereas the non-adaptive 900 kbps-case suffers under large loss. The "received"-curve of non-adaptive 900 kbps lies under the "received"-curve of adaptive session. The reason for this is that this diagram shows the video data amount of the video frames received completely.

The adaptive session driven by the MNC transferred approx. 2.4 times as much data as the non-adaptive 308 kbps-case. This results in a higher video quality as shown in Section VI-B.

B. Video Quality

Figure 5 shows the PVI diagram for the same scenario. The y-axis represents PSNR values. The x-axis shows values for the lengths of frame-freezing due to errors. PSNR values below 25 dB are considered to be unacceptable. For reasons of simplicity, the PVI uses only three quality levels for PSNR. As mentioned before, the threshold for the unacceptable interruptions is set to 1 s, which is considered to be very annoying. Results are plotted in a manner of a plot-box, i.e. the maximum value, upper quartile, median, lower quartile and minimum value of a distribution are shown. The median value is represented by a square symbol. Additionally, the mean value is plotted (as a circle symbol). The x-coordinate of the "plot-box" of PSNR accords to the median value of the interruption-length distribution. For reasons of understandability, we will enter the data in this novel diagram stepwise and begin with PSNR values.

TABLE II
LIMIT VALUES FOR PSNR AND VIDEO INTERRUPTS

session	$p(PSNR <= 25dB)$	$p(interr > 1s)$
308	0.009	0.266
900	0.270	0.356
308-900	0.033	0.138

We see again our three cases. The 308 kbps-session shows the worst median and average of PSNR. The PSNR distribution is narrow (the distance between first quartile and third quartile is small. Between them, 50% of the data lie). The 900 kbps-session has much better median and average. But, the distribution is wide. The probability for unacceptable video quality is 27% (Table II). The adaptive 308-900 kbps-session controlled by the MNC outperforms both non-adaptive sessions (narrow distribution, high median and average, low probability for unacceptable quality) clearly. In this case, the video streaming client correctly receives video frames in much better quality than in the case of the 308 kbps-session

Figure 6 shows the completed PVI diagram. In it, the characteristic values of the video-interruption-length distributions are also depicted as a "horizontal plot-box". Additionally, numbers of samples in these distribution are denoted. The y-coordinate of the video-interruption-length "plot-box" accords to the median value of the appropriate PSNR distribution.

We see that the non-adaptive 900 kbps-session experiences 185 interruptions. The probability of the unacceptable interruption length is 35% (Table II). The maximum interruption value is very large. For this reason, the mean value (black circle symbol) lies above Q_3. In the case of the 308 kbps-session, we experienced only 15 short interruptions. This is understandable, because this video rate was very conservative. The network-driven adaptive session shows 144 interruptions. The probability of unacceptable interruptions is 13%. But most of these interruptions lie marginally above 1 s. To break down large interruptions in a lot of small interruptions, rate shaping mechanisms [2] in the MNC can be used. The resulting "slide show" caused by rate shaping is better than a large video interruption.

VII. Conclusion

In this paper, the idea of the multimedia network controller in the RAN was proposed in order to increase video streaming QoS. The QoS improvement was realized by the network-driven video-content adaptation, where all necessary decisions were made in the RAN. The advantages of this solution are: fast reaction to the changes of the radio link quality, precise decision, simple up-down-rate-switching algorithms, global sight of the cell. The MNC can contain additional elements, for example, elements for rate shaping, which increases QoS in the case of non-adaptive video streaming session or when the video server cannot offer a smaller bit rate any more.

Network-driven solution was evaluated with the help of a novel PVI-diagram allowing to show and compare video quality graphically in a compact form. The adaptive sessions controlled by the proposed MNC outperformed both non-adaptive sessions clearly.

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