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Error resilient video over multimedia broadcast multicast services (MBMS)

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Abstract With available data rates for mobile devices constantly increasing, services such as video broadcast and multicast are becoming feasible. A new standard called Multimedia Broadcast Multicast Services (MBMS) is being developed by 3GPP to enable new class of spectrum-efficient multimedia services. Multicast services are expected to serve a diverse user base with varying connectivity and capabilities. We analyze the problem of video error resilience in MBMS services that is critical to maintain consistent quality for end users. The existing error resilience techniques for IP multicasting are not applicable in the MBMS environment. In this paper, we present error resilience techniques that are applicable within the context of the MBMS standard. We propose an Intra Block Refresh method for MBMS services and the results show improved performance. We develop a methodology that can be applied to adapting traditional error resilience tools for the MBMS environment.

Keywords MBMS · Error resilient video · Multimedia · Multicasting · Broadcasting

1. Introduction

Wireless communication systems have experienced tremendous growth in the last decade and continue to change the way people communicate with each other. A powerful trend is transformation of mobile device from a communication device into an entertainment device and an knowledge resource. With carriers rolling out 3G networks and the mobile devices being capable of accessing WLAN networks, available bandwidth and throughput of mobile devices is increasing. With such data rates, applications such video streaming, audio streaming and broadcast or multicast video are becoming practical on mobile devices. 3GPP (3rd

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Generation Partnership Project) is currently working on a standard for Multimedia Broadcast and Multicast services (MBMS). This standard is expected to be ratified in 2 to 3 years and systems compliant to the standard are expected in 3 to 5 years. We will present an overview of the standard focusing on the proposed architecture, requirements, and on radio aspects.

Due to mobility, the mobile handsets experience wide variety of network conditions such as fading, diversity of signal strength condition, and interference. Video multicasting is seeing renewed interest from mobile service providers where spectrum is scarce and multicasting allows a larger user base. The transmission of video over wireless channels is very challenging especially with real-time requirements in a session shared by multiple users. The members of the multicast / broadcast session are likely to experience diversity of network conditions with varying device capabilities, maintaining the video quality of a multicast session is a challenging problem as this has to consider the needs of all the users in a session. There is a lot of research in the area of video error resilience and a plethora of techniques exist. However, such techniques have typically been applied to applications such as messaging or streaming or video phone. The traditional error resilience techniques for multicasting focused on FEC or retransmission and are applied to the internet or ATM systems. However, MBMS presents different paradigm from traditional multicasting systems due to mobility, diversity of signal conditions, low power and spectrum utilization requirements of receivers.

2. MBMS architecture and concepts

The goal of the MBMS standardization activity is to standardize components and interfaces in the system architecture that would eliminate ambiguity and promote synergy between network operators, content providers, handset manufacturers and network manufacturers [1]. Streaming of live or stored video content to group of mobile devices comes under the scope of MBMS. Some of the typical applications are subscription to Football games, music videos, news clips, weather information and live TV content.

MBMS specifies transmission of data packets from single entity to multiple recipients using a common broadcast channel. This is much more efficient than IP multicast where packets are duplicated for each recipient in a broadcast/ multicast group. The broadcast service is intended for free multimedia services within the broadcast area whereas multicast service requires subscription. Several multicast groups could exist within the broadcast area where multiple sessions could be ongoing at the same time.

Figure 1 shows an example of a multicast broadcast network. This shows the functionalities required for broadcasting, such as Quality of Service (QoS) handling, and where they are implemented within the core network. This architecture enables services offered on the internet or offered by the operator to be broadcast to the terminals.

Figure 2 shows the architecture of GSM / UMTS network that would enable such MBMS services. The Broadcast Multicast Service Controller (BMSC) provides functions for MBMS user service provisioning and delivery to the content provider. It can also serve as an entry point for IP MBMS data traffic from the MBMS User Service source. The GGSN serves as an entry point for IP multicast traffic as MBMS data from the BM-SC. Some of the proposed video error resilience techniques for broadcasting in this paper could be applied either at UMTS Radio Access Network (UTRAN) or GSM/EDGE Radio Access Network (GERAN) or at the server providing service and are depicted in Figure 2.

As Figure 2, depicts Video Resilience adaptation such as increasing intra block rates etc can be applied either at Multicast Broadcast Source (MBS) or either at UTRAN or GERAN. By applying at MBS, it would be compliant with standard compliant however the

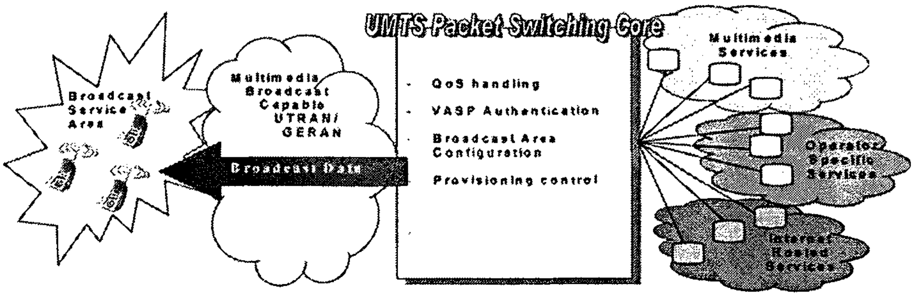

Fig. 1 Example of multicast broadcast mode network

adaptation has to take into account the entire multicast group or groups. A large group tends to have heterogeneous link characteristics and any adaptive schemes becomes quite complex or might not improve performance as the source. A partitioning of the group into multiple groups based on loss characteristics at the source is another possibility however this requires more bandwidth on the infrastructure backbone of wire-line network.

Even though adaptation at UTRAN / GERAN serves better as the group size served within the cell will be smaller (for multicast scenarios, it would in the order of tens of users per channel) but this would require provisioning of adaptation at all GERAN/UTRAN. This would additional cost to the infrastructure provider and is not part of the standard. It also requires the MBS to broadcast high quality video to UTRAN/GERANs. For example, video compressed with all I frames at high quality rate such as 30fps can be sent which UTRA/GERAN can adapt based on their individual multicast group characteristics. This requires more bandwidth on wireline network and computational complexity/cost at the UTRAN/GERANs. However, the adaptation is more effective as loss rates are homogeneous and response time between mobiles and infrastructure is also faster.

We first describe the Multimedia protocols and Codecs specified for use in the MBMS. In the second subsection, we discuss the Radio Interface and relevant requirements.

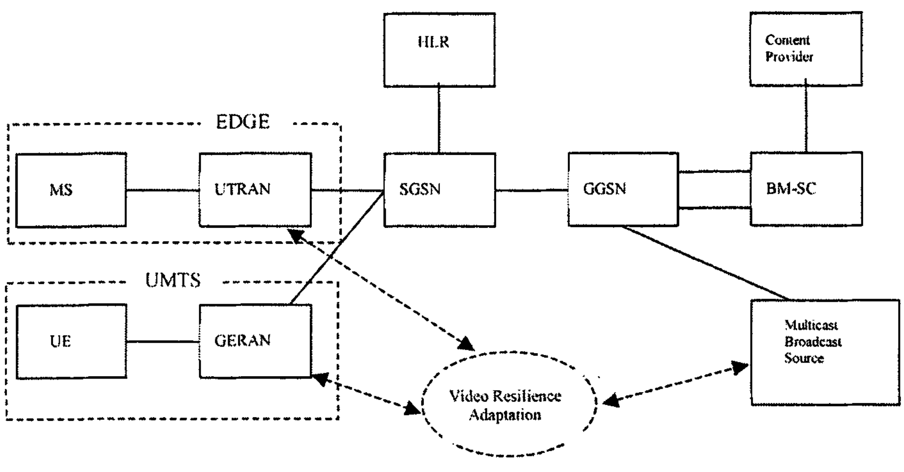

Fig. 2 Architecture of MBMS broadcasting over GSM / UMTS network

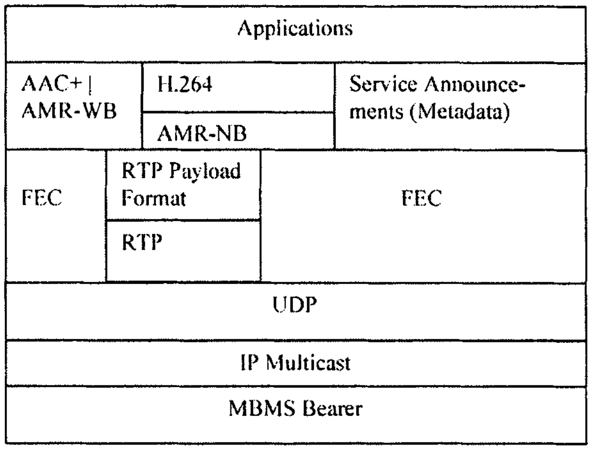

Fig. 3 Software architecture for MBMS streaming

2.1. MBMS protocols and codecs

In this subsection, we give an overview of the MBMS standard focusing on the streaming aspects as they are applicable to multicasting. However, MBMS also standardizes the download and play services aspects that are not addressed in this paper. Figure 3 shows the components involved for MBMS streaming services where we omitted the security related aspects for simplicity. We give an overview of the MBMS protocols, Codecs, FEC model and quality metrics in four subsections.

2.1.1. Streaming delivery protocols

MBMS User Service Discovery/ Announcement are required for advertising MBMS Streaming in advance of, and potentially during, the User Service sessions. MBMS User Service Discovery/Announcement involves the delivery of fragments of metadata to many receivers in a suitable manner. The metadata itself describes details of services. A *metadata fragment* is a single uniquely identifiable block of metadata and could be described as an SDP file. The metadata consists of description of details about MBMS user services, MBMS user service sessions, associated delivery methods and service protection. The data types shall be specified using XML.

The session description protocol (SDP) is also used to describe the multimedia delivery session. A few of the SDP parameters are shown below

- Sender and destination IP addresses
- Transport Session Identifier (TSI) of the session
- Start time and end time of the session
- Media type(s) and fmt-list.
- Data rate using existing SDP bandwidth modifiers.
- FEC configuration and related parameters
- QoE Metrics (described in subsection 2.1.4)

The Real-time Transport Protocol (RTP) is used for delivery whereas RTCP is used for statistics on channel quality. The RTP and RTCP protocols are delivered over UDP/IP protocols. The MBMS bearer services are used to deliver over the MAC and physical layers.

2.1.2. FEC model

There is flexibility from the standard in terms of what FEC scheme to be used. If the selected FEC scheme does not fit, it can be modified. The generic mechanism for systematic FEC of RTP streams specifies two RTP payload formats, one for FEC source packets and one for FEC repair packets with their related signaling. After constructing the source block from the original RTP packets to be protected, the FEC encoder generates the desired amount of FEC protection data, i.e. encoding symbols. These encoding symbols are then sent using the FEC repair packet payload format to the receiver. The FEC repair packets use an SSRC different from the original RTP packets' SSRC, but are sent within the same RTP session. This would avoid non-continuous sequence numbering spaces for both the FEC repair packets and the original RTP packets.

The receivers recover the FEC source packets and places in a buffer for enough to receive the repair packets. The maximum time to store in the buffer is defined as the MIME type.

2.1.3. Media codecs and formats

For voice, AMR NB (Adaptive Multi-rate Narrow Band) Vocoder is supported. For Wideband voice, AMR WB is the mandatory codec for MBMS. The audio is supported by enhanced AAC+ (Advanced Audio Coding) and enhanced AMR-WB. The performance of AMR-WB is strong in low bit rate scenarios whereas AAC+ is strong at high bit rate scenarios. The video codec that is recommended is H.264 (AVC) Baseline Profile Level 1b decoder however H.263 profile 0 level 45 decoder may also be used as it is supported by PSS.

2.1.4. Quality metrics

The MBMS Quality of Experience (QoE) metrics feature is optional for both MBMS streaming server and MBMS client, and shall not disturb the MBMS service. An MBMS Server that supports the QoE metrics feature shall activate the gathering of client QoE metrics with SDP and sends as per reception reporting procedure. Following are the QoE metrics

- Corruption duration metric
  - Time between last good frame before corruption to the first subsequent good frame or end of the reporting period
- Rebuffering duration metric
  - Time during which there is stall in playback
- Initial buffering duration metric
  - Time from receiving the first RTP packet until playing starts
- Successive loss of RTP packets
  - Indicates number of RTP source packets lost in succession
- Frame rate deviation
  - Indicates playback rate information deviatingfrom predefined value

- Jitter duration
  - Absolute difference between actual playback time and expected playback time that is greater than 100 ms.

In MBMS reception reporting will be done only once at end of streaming and all the Jitter duration is summed up as *TotalJitterDuration* and total number of individual events called *NumberofJitterEvents* are reported. The SDP syntax is used for reporting the QoE metrics.

As shown above, the MBMS standardizes critical components such as Protocols, Codecs. However, the actual FEC schemes to be implemented are kept open to promote innovation. Despite the FEC schemes, the wireless channels do introduce errors and require error concealment schemes or robust video codecs that can recover from errors. In this paper, we would concentrate on the radio interface between mobile terminal and RAN (radio access network) where the packet loss in the network could be significant and require appropriate design to prevent degradation in quality.

2.2. MBMS radio access network

The RAN is interface between UE (User Equipment) and Base Station (UTRAN or GERAN) that is the wireless segment and is the weakest link in the entire network. This interface has to be designed appropriately to maintain overall quality. The requirements of this network are especially important in multicasting where several users with heterogeneous error conditions share the same session. The MBMS specified requirements for RAN (Radio access network) are specified here [2].

- MBMS does not recommend retransmissions at the link layer due to losses or upon feedback from the terminals. It however does not preclude periodic repetitions of the MBMS content based on operator or content provider scheduling of retransmissions based on feedback at the application level.
- MBMS session should stay continuous during handover and the requirements specify minimal data loss.
- MBMS also require joining an ongoing session as it would require interoperability with other services.
- MBMS does not specify QoS (Quality of Service) at the RAN layer but expect a certain level of reservation for the service.
- MBMS specifies shared dedicated resources (point to point) or common resources (point to multipoint) and is left to the operator to select connection type. This is based on downlink radio resource environment such as radio resource efficiency.

Based on such requirements, the appropriate error resilient techniques can be determined that would allow efficient use of the spectrum for quality video reception. This would require innovation of new error resilient techniques in this area.

3. Error resilience in Wireless Video transmission

Video communication has substantially higher bandwidth requirements compared to voice communication. However, video data comprises of spatial, temporal and statistical dependencies. These are exploited in video compression techniques to reduce the bandwidth required for Wireless video communication. In the widely used video compression standards such as H.263+, MPEG-4 and H.264, hybrid coding techniques are used where both temporal and spatial redundancy is minimized to achieve higher compression. In temporal prediction techniques, a frame is either coded with no other frame dependency (Intra Frame) or coded with dependency on past frame (P Frame) or coded with dependency on past and future frames (B Frame). Depending on the framing structure where a combination of P and B frames are repeated periodically between two I Frames, the frame dependency exists between two Intra frames. During the transport of video packets or for any other reason, if there is loss of data, the error is propagated until an error free Intra frame is decoded [4].

Such errors can be roughly classified into two categories: random bit errors and erasure errors [3]. Depending on the coding methods and affected information content, the impact of random bit errors can range from negligible to objectionable. Erasure errors, on the other hand, can be caused by packet loss in packet networks. To combat such errors, two commonly used error resilience techniques are techniques are Forward error correction (FEC) and ARQ strategies. The FEC techniques add certain redundancy that is used to detect and correct errors. However, the amount of redundancy that is added is independent of the receiver's current conditions; it is redundant in good signal strength conditions and ineffective in very bad signal strength conditions. The amount of FEC required in the case of multicasting is even difficult due to heterogeneous conditions of the receivers. The ARQ strategies that rely on retransmission are adaptive to the signal strength conditions however the delay is unbounded. A combination of FEC and ARQ strategies are deployed as Hybrid ARQ techniques during retransmission the parity packets are sent or perform retransmissions when FEC is not sufficient. The ARQ strategies are not recommended in MBMS systems with a large group where there could be feedback implosion.

Despite the ARQ and FEC strategies, the wireless networks would experience bit errors so the video decoder should be capable of error concealment to compensate or mitigate errors. The application should be able to minimize the impact of error by localization and prevention of error propagation. The error concealment techniques are also crucial in the case of multicast applications where errors are unavoidable in heterogeneous conditions. The FEC or ARQ schemes are independent of the data and can be applicable to any type of data. However, we use the term *video error techniques* to refer to techniques that are specific to video data by utilizing the knowledge about how the video is compressed and packetized.

The video error resilience techniques are classified into the following

- Encoder based techniques
- Decoder based techniques
- Interactive based techniques
- Proxy based techniques

An overview of video error resilience techniques is presented in [3]. Error resilient support provided in H.263+ or MPEG-4 standards is discussed in detail in [2]. The H.264 standard and a survey of wireless error resilience techniques for streaming and messaging applications are described in [5].

A survey of error resilient techniques for multicast application for IP-based networks is presented in [6]. The paper presents algorithms that combine ARQ, FEC and local recovery techniques where the retransmissions are conducted by multicast group members or intermediate nodes in the multicast tree. With heterogeneous receivers experiencing several error rates, different multicast groups are formed. Video error resilience techniques using hierarchical algorithms are proposed where transmission of I, B and P frames are sent with varying levels of FEC protection. The RTP (Real-time transport protocol) payload formats for transporting UDP packets together with RTCP (Real-time control protocol) feedback reports is also utilized in multicast networks.

Error resilience support for multicast and broadcast services in CDMA 2000 as well as the algorithms for efficient bandwidth utilization are studied in [7, 8].

Some of the prior research work on error resilience for broadcast terminals focuses on increasing FEC based on the feedback statistics from the users [9]. The additional FEC protection is based on the average error rate experienced by members of the session. Also, research on proxy based error resilience is presented in [9]. However, the proxy requires individual retransmission based on feedback from the terminals. [10] presents comparison of different error resilience algorithms for wireless video multicasting on WLANs (Wireless Local Area networks). [11] presents novel methods for multicast and unicast real-time video streaming over wireless LANs using hybrid ARQ techniques. The multicast system takes into account the heterogeneity of receivers and uses that information to minimize distortion experience to each user. However, in a literature survey we found that error resilient techniques at the video codec level are not applied to multicasting areas. Our work explores the use of video error resilience tools for multicast applications.

4. Methodology of video error resilience techniques for MBMS

The MBMS systems present a new paradigm from the traditional internet or satellite based multicasting system where due to mobility, the system has to account for wide variety of receiver conditions such as handover, speed of the receiver, interference and fading. Apart from that, the required bandwidth and power should be kept low for the mobile devices.

The interactive and proxy based techniques where lost packets are requested by the receiving terminal for retransmission are ruled out in MBMS systems. The ARQ or FEC strategies also cannot guarantee the wireless transmission within multicasting time constraints. After error correction, even if there is one error it is declared as packet loss by the packet schemes. Also, due to the heavy compression in video codec and variable length codes used for entropy coding of coefficients, a single error could render the packet undecodable. However, some parts of the packet can still be decoded if the error can be localized. The error localization techniques are discussed in [12]. The tools that MPEG-4 standard offers for error resilience are shown below.

- Resync markers
- RVLC (reversible variable length codes)
- Header retransmission

The resync markers are patterns that are placed within the encoded bit stream. A packet could contain multiple resync markers. If the decoder detects an error, it can scan for the next resync marker. Upon scanning a valid resync marker, it can start decoding again. This would enable recovery of partial packets instead of losing entire packet.

The property of RVLC is, with a little coding overhead, the coefficients could be decoded both ways. Upon detecting an errors and jumping to the next resync marker, the decoder can start decoding backwards thereby giving an opportunity to recover the packet as much as possible.

In a multicasting session, the header retransmission is useful in multicasting or broadcasting when members can join the session at any time. It is also expected that during the session setup, they would be given the header information necessary to join the session. These techniques are quite useful in error concealment in multicasting sessions.

When a mobile joins an existing broadcast session, there is a delay before which it can get synchronized. This delay is proportional to the frequency of I frames as determined by the system. Since I frames require more bits than P frames, the compression efficiency is inversely proportional to the frequency of I frames. If f is the frequency of the I frames and r is the frame rate of the video compression, the worst case start delay in seconds,

$$D = (\mathrm{f} - 1)/\mathrm{r}.$$

The application would also require more frequent transmission of I frames so as to allow the user to join an ongoing session. We expect the broadcasting session to have low frequency of I frames otherwise there is a perceivable delay in joining the session. But the use of frequent I frames would result in higher bandwidth. Another alternative is that during session admission, an I frame is unicast to the member. Such scheme would make startup delay independent of I frame frequency. The frequent I frames are a good tool against error propagation provided they can be transmitted error-free.

One other tool that is effective against error propagation is Intra block refresh technique. Periodically, depending on the algorithm, certain percentage of P or B frames blocks are coded as Intra blocks as intra blocks terminate error propagation. However the intra block refresh technique is more useful if the frequency of I frames is lower. Typically the percentage of Intra blocks is fixed. Varying the percentage based on the signal strength conditions will lead to better video quality and end user experience.

In multicasting scenarios, using RTCP feedback reports, the average signal strength conditions of the group members can be determined. Based on that, the percentage of intra blocks can be determined. If the receivers are experiencing high error rates, the intra block percentage is increased and vice versa. This technique is effective for a smaller group of members such is the case in a wireless cell served by a base station. We foresee that a proxy would integrated with the base station that handles the multicasting and has the capability to vary the intra block percentage through decoding and re-encoding video stream coming from the sender. Based on the carrier preferences, different strategies could be deployed. The intra-block percentage is determined based on the group statistics.

The optimization of overall PSNR can be based on criteria that can be driven by the service provider or the equipment manufacturer. One goal would be to provide minimum quality to all receivers whereas another one would be to increase overall quality without penalizing any users significantly. If certain members experience severe losses, guaranteeing minimum quality could mean transmitting excessive repair packets to the entire multicast group. This could be undesirable to the receivers as it would impact their battery life or increase billing. Having an additional channel for repair packets would reduce this impact however this could require multiple receivers on the handset that adds cost and battery life to the system. The second approach of increasing overall quality without adversely penalizing users is a good approach which is what selected for our experiments. Similar approach was undertaken in prior art [12].

5. Experimental setup, results, and discussion

An MBMS delivery environment was simulated with a given packet loss rate. The burst errors caused by fading are treated as packet losses. Since packet data network is used for transport of video over wireless networks, any errors despite error correction would result in packet loss. Since broadcasting involves several independent links that are uncorrelated, the losses in receivers from the Base station perspective can be assumed to be random. The Foreman sequence was encoded using MPEG-4 SP, QCIF, with a bit rate of 64 kbps and 15fps. An

I frame is repeated every 5 seconds and note that this value determines the synchronization time for joining members. The sample size is 100 mobiles with baseline fixed IBR at 5%. The range of loss percentage was chosen from 1 to 24% since losses over 25% result in severe quality degradation and result in user dropping the session.

The results are evaluated by dropping packets as determined by the loss model and decoding the remaining packets, and recording the PSNR values at each mobile. The error concealment technique performed by the decoder depends on the way frame was encoded. For an I frame block, the dc component of the lost macroblock is calculated from the neighboring blocks. For a PVOP, the concealment steps in calculating MV's is outlined in H.263 Annex F are undertaken.

The average PSNR for the group is used as the metric. We found that if the loss rates are heterogeneous, varying the IBR has little impact on the overall PSNR. In the second baseline, the users were divided into groups where each group has a packet drop rate in the ranges of 0–8%, 9–16% and 17–24%. This would mean having three multicast groups instead of one multicast group within a cell and requires more bandwidth. The alternative is to use the same bandwidth for one multicast group to send more repair packets resulting in improved quality. However, this is unfair to mobiles that experience good signal strength conditions in two aspects: 1) extra power required in decoding repair packets that are not necessary and 2) extra cost in packet based billing systems. Partitioning the users according to their loss characteristics also allows better user management and improves the QoS by using an IBR rate appropriate for the group.

Table 1 shows the results of the experiments with varying group size and loss rate. IBR percentage is varied and the three sets of experiments are conducted. Since the exact loss in quality cannot be estimated from a lost packet, 30 trials are run for each point and the average PSNR is recorded. From the table, based on group average loss rate, the IBR rate based on the knowledge of packet loss conditions can make significant impact on the average PSNR. The PSNR reaches a peak for a particular IBR rate and this varies depending on the loss rate. For example, 0–7% loss rate, from the results a peak is seen around 3% IBR rate. For 16–24% loss rate, the peak PSNR is found at 15% IBR. In a nutshell, the loss in error-free quality due to Intra blocks is overcome by the gain in error resilience upto a specific IBR rate and after certain rate, the performance of IBR seems to be flat. At this point, as IBR rate is increased, it would result in loss in quality during encoding. This "peak" IBR rate varies depending on the loss rate. The average PSNR can be improved by determining the "peak" IBR rate for group of homogenous loss rates.

In the above case, for loss rate range of 0-25%, three groups were assumed. For 5 groups with each group ranging within 5% percentage, the performance is more as loss rates are homogeneous however this requires more bandwidth.

As shown in the figure 4, the peak average PSNR varies depending on the loss rate. For the first group with loss rate between 0-5%, the peak PSNR corresponds to IBR rate of 5% where as for fourth group with loss rate between 15-20%, the peak PSNR corresponds to IBR rate

Table 1 Effect of increasing IBR percentage for various loss rate percentages (I frame frequency=5sec, bit rate= 64 kbps, 3 groups, Random model)

	0–7%Loss	8–15% Loss	16–24% Loss
0%IBR	25.71	20.77	18.82
3%IBR	***26.72***	23.13	20.85
10%IBR	26.00	***23.54***	21.3
15%IBR	25.63	23.25	***21.46***
25%IBR	25.34	23.14	21.42

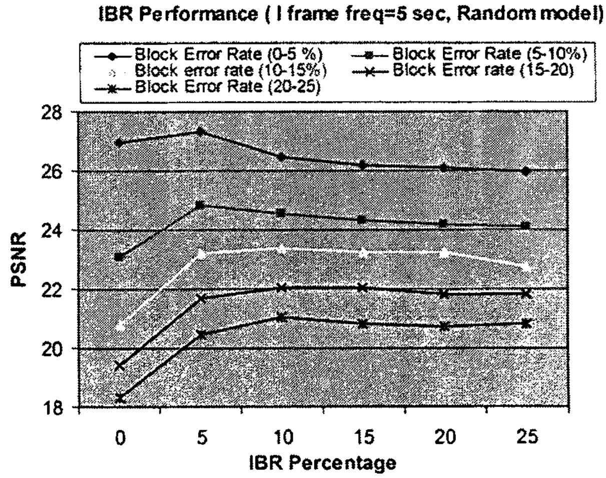

Fig. 4 Average PSNR for varying IBR rate for 5 groups (I frame frequency=5sec, bit rate=64 kbps, 3 groups, random model

of 15%. At higher loss rates, the IBR rate effect seems to saturate where even though IBR rate is increased, the PSNR stays around the same level as shown for fifth group (20–25%). Within the same context, two groups were devised with loss rates ranging from (0–12%) and (12–24%) and still the varying the IBR rate does improve the performance however it is not as prominent as 3 or 5 groups. Figure 5 depicts the performance of such partitioning.

The loss percentages can be modeled as Gaussian random processes with mean μ and variance σ and for modeling groups with such loss percentages, the average PSNR improves. From the single user studies, for a particular loss rate, an IBR rate can be determined that results in optimal PSNR. Similarly, for multicast groups, if the IBR percentage is determined based on the mean loss rate, the performance can be improved. With such modeling, the loss rates of the group can be characterized as forming a Gaussian group, an optimal PSNR can be

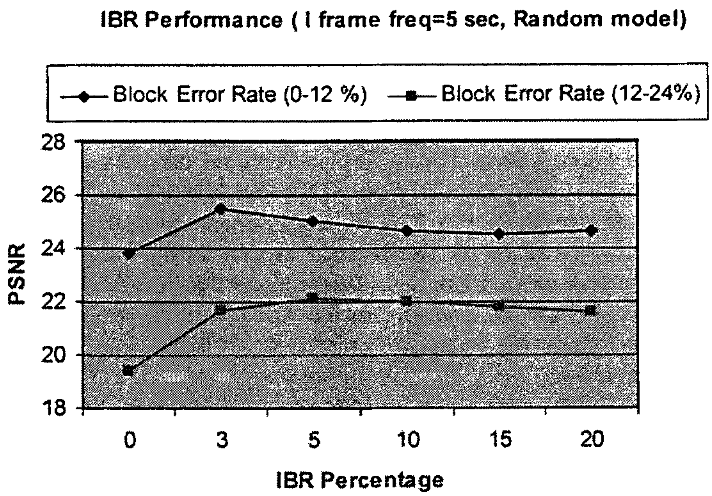

Fig. 5 Average PSNR for varying IBR rate for 5 groups (I frame frequency=5sec, bit rate=64 kbps, 2 groups, random model)

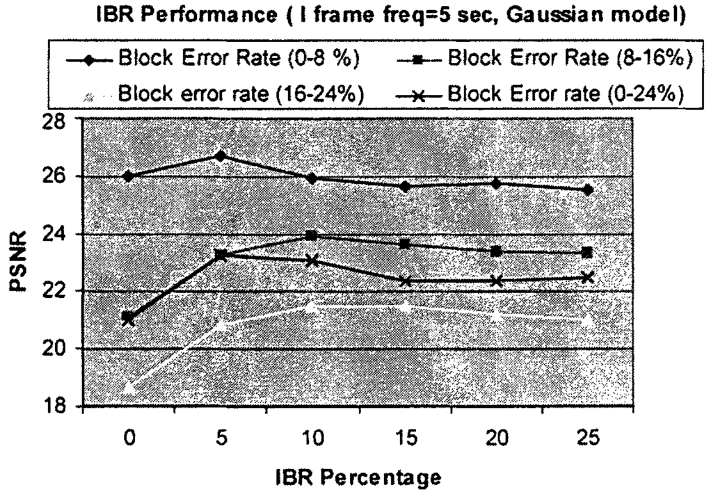

Fig. 6 Average PSNR for varying IBR rate for 3 groups and single group (I frame frequency=5sec, bit rate=64 kbps, 2 groups, random model)

obtained for heterogeneous groups. Such classification could be close to realistic assumptions of multicast scenarios.

For such modeling, the performance improvement due to such partitioning results in about 0.77 db performance improvement over no partitioning as shown in Figure 6. However, still a peak PSNR could be still found for heterogeneous loss rates. This would mean having three multicast groups instead of one multicast group within a cell and that requires more bandwidth. In some cases, the same bandwidth could be used in one multicast group to send more repair packets resulting in lower packet loss. However, this is unfair to mobiles that experience good signal strength conditions in two aspects. One aspect is the extra power required in decoding repair packets that are not necessary. This would impact battery life that is big concern in mobile devices. Another aspect is billing by carrier wherein the charges are based on the number of packets received. The extra repair packets needed for a worst case user can result in extra charges for the receiver. In this context, we think it is necessary to devise schemes that take into account the device constraints.

For the case of deterministic random modeling, the mobiles have a fixed loss rate. The total number of repetitions is divided by the loss rate range and the result is used to determine the number of mobiles having same loss rate. This modeling is to reduce variability and resulted in much better results. Figure 7 shows the performance of such technique. In such modeling, even a single group shows better peak. As shown, an I frame frequency is also varied for 5 & 10 seconds and for both cases a distinct peak PSNR is found. The simulations were performed for very low bit rate cases such as 48 kbps and high bit rates such as 384 kbps for QCIF resolution.

The above table shows that as grouping of users into multicast groups based on their loss rates and varying IBR results in different qualities. Since mobility is expected during session, there is typically huge packet loss during handover. If the packet loss occurs during I frame, the effects are severe. In the case of P frames, the error concealment techniques could mitigate the loss however the distortion would continue to propagate until I frame is found. This is also combated using intra block refresh rates. The loss of B frames is negligible and limits the loss to that particular frame.

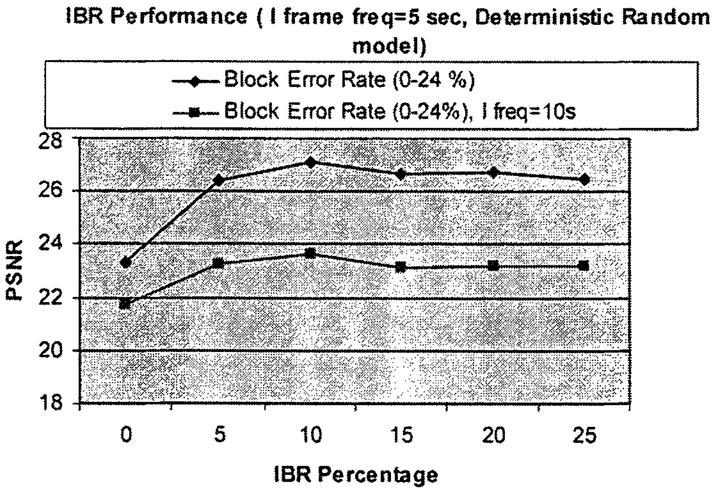

Fig. 7 Average PSNR for varying IBR rate for single group (I frame frequency=5sec, bit rate=64 kbps, single group, Deterministic random case)

The RTCP feedback reports could be reported to the sender and application based retransmission of packets is not precluded in MBMS. However, the effectiveness of such technique depends on round trip delay between the sender and the receiver. Since multicast sessions are shared, it could results in waste of bandwidth. In MBMS, a point to point retransmission could be applied however this is used in extreme losses such as I frame loss as it is bandwidth intensive.

5.1. Prediction structure for improved error resilience

Apart from the traditional multicasting / broadcasting techniques, the MBMS system requires new techniques for error resilience. Since MBMS does not allow for retransmissions, the temporary fading conditions of wireless channels could result in corruption of certain frames. Due to the frame dependency within the hybrid coding techniques, the errors propagate until an I frame is decoded. In our prior work, we propose Periodic Intra Frame Based Prediction (PIFBP) structure in which the frames are based on the same I frame and corruption of a P frame would not affect the next P frame [15]. The frame structure is shown in Figure 8. Figure 9 depicts the performance of PIFBP scheme with the performance of MPEG-4 Simple Profile (SP) codec during frame loss. The percentage of frames that are dropped is varied and it is clearly seen that PIFBP scheme maintains the quality whereas the MPEG-4 (SP) codec with varying I frame periods, the quality degrades with increase in frame loss percentage.

H.264 includes syntax for prediction from previous frames and this technique compliant with the H.264 standard. In the experiments, I frames were not dropped but on the P frames were dropped. This would require significant bandwidth for protection of I frames such as FEC or ARQ. There are other advantages apart from error resilience as this could improve synchronization time. Any mobile joining the multicast session can be unicast to an I frame and upon decoding, it could join the multicast session as P frames would be based on that I frame. This is not possible in traditional schemes as mobile has to wait till the start of GOP period. However, any error in I frame would effect the entire GOP period and hence requires ARQ and/or extra FEC for protection. By using feedback schemes, the I frames or repair

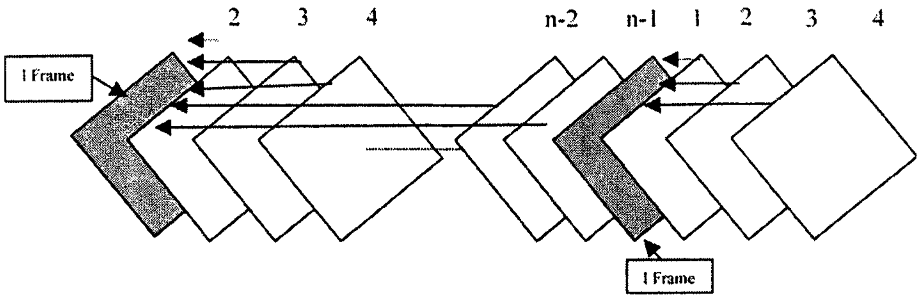

Fig. 8 Periodic intra frame based prediction

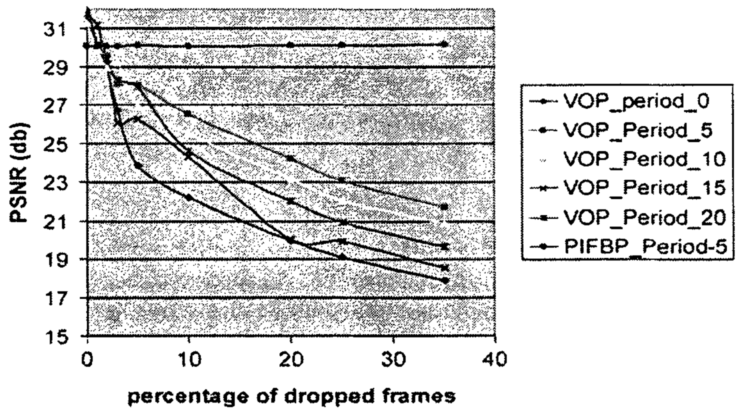

Fig. 9 Performance under loss of Inter / Intra frame coding (Data rate=128 kbps, fps=15 fps)

packets could be unicast to the mobiles. An adhoc network among peers could be used to recover the I frames thereby reducing the core bandwidth limitations. The investigation of such schemes is currently underway.

6. Conclusions

We presented an overview of the emerging MBMS standard. The MBMS standard is expected to enable new class of multimedia services to mobile devices. The video services delivered over MBMS require new approaches to error resilience. The traditional network level error resilience techniques developed for IP multicasting are not suitable and sufficient for MBMS services. We developed a methodology for error resilience in MBMS network that attempts to maximize the quality of experience for the multicast users as a group. The well-known intra-block refresh technique is used to demonstrate the methodology. The quality of experienced by partitioning the users dynamically based on their loss characteristics. We showed that partitioning users into groups based on their loss rates increases the overall quality. We also present a robust video structure (PIFBP) to improve the error resilience of video services over MBMS.

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