

# E-LETTER

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## CONTENTS

Message from Editor-in-Chief .....	2
<b>SPECIAL ISSUE ON ADVANCES IN IPTV .....</b>	<b>3</b>
<b>IPTV – A New Platform for Information and Entertainment in the World of Digital Evolution .....</b>	<b>3</b>
<i>Guest Editor: Bin Wei, AT&amp;T Labs – Research, USA .....</i>	<i>3</i>
<b>Making IPTV Robust and Scalable: A Service Provider's Perspective .....</b>	<b>5</b>
<i>Kadangode K. Ramakrishnan (IEEE Fellow), AT&amp;T Labs – Research, USA .....</i>	<i>5</i>
<b>Network-Reconvergence Technologies for Low-loss Video Transport.....</b>	<b>11</b>
<i>John Evans, Ali C. Begen, and Jason Greengrass, Cisco, USA .....</i>	<i>11</i>
<b>KreaTV in focus: The settop box and beyond.....</b>	<b>17</b>
<i>Martin Bengtsson, Motorola, Sweden.....</i>	<i>17</i>
<b>Is IPTV a quality experience? Quality monitoring in an IPTV network .....</b>	<b>20</b>
<i>Amy R. Reibman (IEEE Fellow), AT&amp;T Labs – Research, USA .....</i>	<i>20</i>
<b>Serving Personalized Content in IPTV Platforms .....</b>	<b>22</b>
<i>Zhu Liu, AT&amp;T Labs – Research, USA .....</i>	<i>22</i>
<b>. 3D Games Deployment Platform for IPTV .....</b>	<b>25</b>
<i>Abdenmour El Rhalibi, Madjid Merabti, Liverpool John Moores University, UK ....</i>	<i>25</i>
<b>TECHNOLOGY ADVANCES .....</b>	<b>39</b>
<b>Editor's Selected Book Recommendation.....</b>	<b>39</b>
<b>Editor's Selected Paper Recommendation .....</b>	<b>41</b>
<i>Focused Technology Advances Series .....</i>	<i>42</i>
<b>Enabling Broadcast of User-Generated Live Video without Servers .....</b>	<b>42</b>
<i>Chao Liang and Yong Liu, Polytechnic Institute of NYU, USA.....</i>	<i>42</i>
<b>IG Corner: Seamless Quality of Service Interworking across different Access Technologies .....</b>	<b>45</b>
<i>Apostolis K. Salkintzis, Motorola, Greece .....</i>	<i>45</i>
<b>MMTC COMMUNICATIONS &amp; EVENTS .....</b>	<b>48</b>
<b>Call for Papers of Selected Journal Special Issues .....</b>	<b>48</b>
<b>Call for Papers of Selected Conferences .....</b>	<b>49</b>
<b>Next Issue Partial Content Preview.....</b>	<b>50</b>
<b>E-Letter Editorial Board.....</b>	<b>51</b>



IEEE COMMUNICATIONS SOCIETY

## IEEE COMSOC MMTC E-Letter

### Message from Editor-in-Chief

In this Issue, we feature a Special Issue on Advances in IPTV, put together by the Guest Editor, Dr. Bin Wei (AT&T Labs Research, USA). Six articles were invited in the Special Issue from world well-known professors and scholars to address various technical foci of the IPTV technology. Dr. Wei provide us more details about this Special Issue in his Editorial article on page 3.

In addition, we feature a new column, Editor's Selected Book Recommendation, to provide book reviews for books with careful selection. The goal of this column is to introduce new books to our fellow members with opinions and recommendations from the Column Editor. This time the selected book recommended by our Column Editor, Dr. Shiguo Lian (France Telecom R&D, China) is "*Multimedia Networking: From Theory to Practice*" written by Dr. Jenq-Neng Hwang. Please check out more details in the article.

In this issue, the selected paper for recommendations by our Column Editor, Dr. Guan-Ming Su (Marvell Semiconductors, USA), is an article recently published in *IEEE Journal of Selected Topics in Signal Processing*, in which a novel video quality assessment scheme is proposed for video communications applications.

In the focused technology column, Mr. Chao Liang and Dr. Yong Liu (Polytechnic Institute of NYU, USA) introduce an alternative way for server-based live video broadcasting, that is, a "pure" P2P streaming solution without any server support. Check out this article for more details.

In the IG column, Dr. Apostolis K. Salkintzis (Motorola, Greece) discusses one of the major focused topics of the QoS Interest Group (QoSIG): seamless QoS interworking across different



access networks for multimedia communications.

Again, I would like to take this opportunity to express my deep personal thanks to the Guest Editor, Dr. Bin Wei, who has been working very hard around clock to make this Special Issue a great success. I would also encourage our fellow members to propose new Special Issue in the coming E-Letters on other topics to educate us the latest advances as well as ideas and opinions from different perspectives.

As always, I thank all Editors of the E-Letter, and our authors to make this issue successful.

Thank you very much.

Haohong Wang  
Editor-in-Chief, MMTC E-Letter

## IEEE COMSOC MMTC E-Letter

### SPECIAL ISSUE ON ADVANCES IN IPTV

#### **IPTV – A New Platform for Information and Entertainment in the World of Digital Evolution**

*Guest Editor: Bin Wei, AT&T Labs – Research, USA  
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IPTV, which stands for Internet Protocol Television, is an aggregation of new technologies in computing, networking and storage to deliver high quality TV content along with a rich set of services through IP-based networks. IPTV fundamentally breaks the barrier of one-way communication inherited in traditional broadcast TV services and creates a platform which brings information and entertainment as part of each other to a new level.

With this IPTV special issue, we hope to provide readers a brief outlook of what the IPTV is all about. We have selected 6 papers, to cover a wide spectrum of IPTV topics from media generation all the way to media consumption, including service provider's perspective, equipment provider's perspective, user's perspective, emerging application's perspective, and issues on user generated content.

For readers who like to know how a typical IPTV network is constructed out of existing IP networks for IP-based video distribution, you may find a brief description and further references in Dr. Ramakrishnan's paper "Making IPTV Robust and Scalable: A Service Provider's Perspective". In addition to the IPTV network structure, the paper also discusses several major issues in building and using such a network.

The paper "Network-Reconvergence Technologies for Low-Loss Video Transport" by John Evans, Ali C. Begen and Jason Greengrass, presents their ideas of video quality control from network transport perspective. By combining network-level technologies and application-level approaches, the paper improves video transport for the traffic over IP/MPLS networks.

Discussing IPTV networks without some detail about set-top boxes is incomplete, because they are the devices that bring all IPTV functions to customers. Martin Bengtsson's paper "KreaTV in focus: The Set-top box and beyond" depicts a picture of how powerful a set-top box can be by using KreaTV as an example. For those who

would like to integrate more functions to set-top boxes, you may find some hints from the paper.

High quality video is essential for IPTV as a new TV related service compared with the conventional TV. IP network impairment can affect video quality in various ways. Thus a well designed video quality monitoring system is indispensable for a reliable IPTV service. In the paper, "Is IPTV a Quality Experience? Quality Monitoring in an IPTV Network" Dr. Reibman presents some key points in the framework of video quality control.

The merit of IPTV is reflected by the potentials of integrating more services and applications, such as personalized content, games, or digital entertainment, which are not feasible in the traditional broadcast TV platform. In his paper, "Serving Personalized Content in IPTV Platforms", Dr. Liu describes an appealing service which helps customers to create their own channels. The paper also presents some perspectives in addressing challenges of personalization services in IPTV systems. Dr. El Rhalibi and Prof. Merabti point out the limitation of current approach for gaming applications on IPTV and propose a new gaming framework in their paper, "3D Games Deployment Platform for IPTV". The paper also presents the design, development, and deployment in great detail.

Hopefully by reading these papers, readers can obtain a comprehensive image about the present and the future of IPTV.

On the other hand, there are also research efforts for video distributions without relying on dedicated servers. The paper by Chao Liang and Prof. Yong Liu, "Enabling Broadcast of User-Generated Live Video without Servers", in Focused Technology Advances Series, is an example of approaches which essentially rely on P2P support.

With that, I'd like to thank all the authors for their contributions.

## IEEE COMSOC MMTC E-Letter



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**Making IPTV Robust and Scalable: A Service Provider's Perspective**

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With the widespread availability of broadband connectivity to homes, IP-based distribution of real-time television, IPTV, is now a reality. The term IPTV refers to an array of video services (including “linear-TV”, i.e., real-time programming, and video-on-demand) where the video is compressed, and transmitted to subscribers over an IP packet data network. Carriers are implementing IPTV that streams real-time content using IP multicast. Consumers expect the delivery of video over IP to meet the same quality and reliability characteristics of other (non-IP based) options that are available for video delivery, such as over cable and satellite. Carriers deploying IPTV seek to scale the distribution so that it is available nationwide to a very large number of consumers (subscribers). In addition, to compete with alternate services, they provide a comparable service offering hundreds of channels of “linear-TV” as well as video-on-demand with a large content library. Thus, the scale and robustness requirements for IPTV requires us to design the IP network with the right combination of efficient IP-based distribution of video content, robust network restoration, and video and packet recovery methods.

IPTV places significant bandwidth demands on the infrastructure. With a typical SD-quality channel consuming 2 Mbps and an HD-quality channel consuming 6-15 Mbps, the IP infrastructure needs to be cost-competitive with the more mature cable broadcast infrastructure. IP Multicast (Protocol Independent Multicast (PIM) [1,2,3]) enables carriers to distribute the content from the source to the various distribution points on the IP backbone efficiently, as a packet traverses each link only once. Some carriers have chosen to use PIM-Dense Mode, where the video traffic is first flooded throughout the backbone network, and the tree is “pruned” back along branches where the traffic is not wanted. The underlying assumption is that there are multicast receivers for this group at most locations, and hence flooding is appropriate. The use of PIM-Dense Mode may be justified based on the intuition that a particular metropolitan

area is likely to have at least one subscriber for each “linear-TV” channel that is being distributed. However, this can result in considerable overhead (as the traffic would be flooded until it is pruned back) if extended beyond the backbone. Every router also has to keep state for the multicast group. Thus, other carriers have chosen to use “PIM-Sparse Mode” (PIM-SM) for wide scale deployment of IP multicast for both densely and sparsely populated groups. With PIM-SM, video traffic is sent only where it is requested, and receivers are required to explicitly join a multicast group to receive traffic. In particular, for a nationwide distribution of linear-TV, PIM-Source Specific Multicast (PIM-SSM) has been adopted.

It may be useful to see the general network architecture and design for distribution of broadcast video and other services. Figure 1 [4] shows as an example, a simplified network topology for segments that deliver IPTV service in the continental US. The Super hub office (SHO) gathers content from the national video content providers, such as TV and cable networks (mostly via satellite today) and distributes it to a large set of receiving locations, called Video hub offices (VHOs). Each VHO in turn feeds a metropolitan area. IP routers are used to transport the IPTV content in the SHO and VHOs. The combination of SHO and VHO routers plus the links that connect them comprise the long-distance IPTV backbone. The VHO combines the national feeds with local content and other services and then distributes the content to each metro area. The long-distance backbone network between the SHO and the VHO includes a pair of redundant routers that are associated with each VHO. This allows for protection against router component failures, router hardware maintenance, or software upgrades. The links of the long-distance IPTV backbone are usually 2.5 or 10 Gbps Ethernet or SONET. The aggregate bandwidth requirement for all the television channels (several hundred and growing) carried in the backbone can be a significant (over 60-70%) proportion of the individual link capacity.

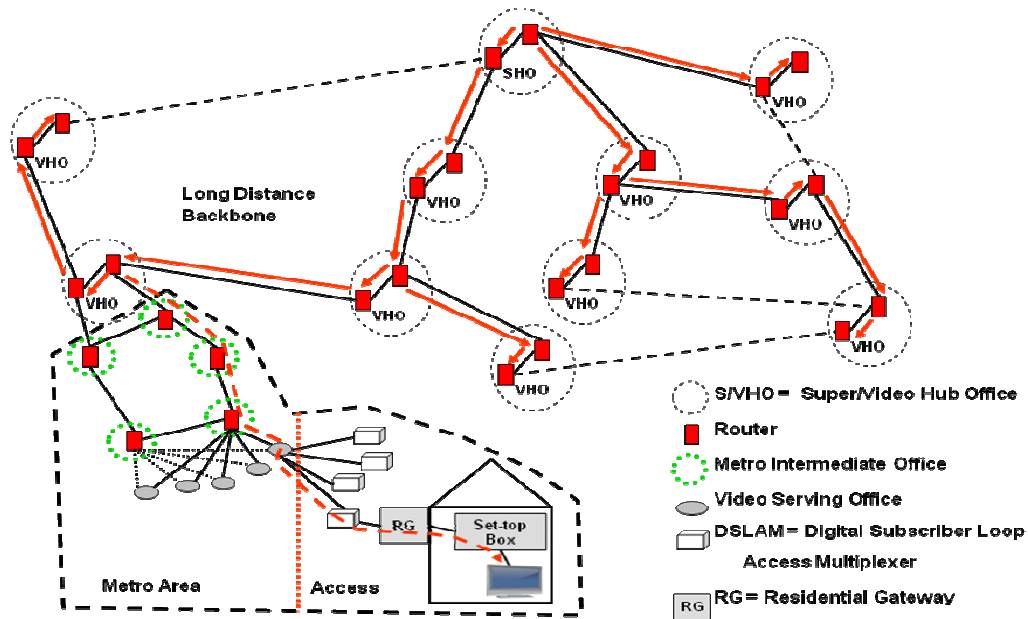


Figure 1: Example nationwide IPTV network

Our group has been working on several aspects related to IP-based video distribution. We have focused on ways to enhance the reliability and scalability of the IPTV architecture, protocols and mechanisms. We have examined ways of restoring connectivity using a combination of MPLS Fast Re-route and OSPF/PIM reconfiguration even after multiple link failures [4]. When a user switches from one channel to another, IPTV typically resorts to a short period of unicast of the new channel at a faster rate so that the user begins to view the new channel quickly. However, such approaches may result in scalability concerns when the number of users concurrently changing channels goes up. We have developed an approach to continue to use multicast, while still achieving fast channel change [5]. Finally, video-on-demand is a growing segment of video-distribution both over traditional channels as well as over IP networks. The popularity of on-demand content, including TV shows that are available on the Internet suggests that it is going to continue to grow. To meet the scalability requires for large-scale video-on-demand, in terms of user population, the need to adjust varying request arrival patterns including “flash crowds” of a large burst of users accessing one or more popular pieces of content, and to accommodate a large content library, we have developed an architecture for provider-based delivery of VoD using Cooperative Peer

assists and Multicast (CPM) [6]. In this article, I will briefly cover each of these, and refer the reader to the corresponding publication referenced for more details.

### Designing an IPTV Network Backbone for Reliability

It is important that failures in the IPTV backbone network do not result in user perceived impairments. Although failures in the network are infrequent, they are not rare in comparison to typical carrier requirements to achieve 99.999% availability (typically termed “5-9s”, resulting in no more than 5 minutes of disruption per year). The vast majority of network (physical layer) failures affect only a single link, which can be restored using link-based fast reroute (FRR) [7] within roughly 50 milliseconds. Link-based FRR creates a “pseudo-wire” or tunnel in parallel to the IP adjacencies (links) along the forwarding path used by the PIM-SSM tree. For each such tunnel both a primary and backup path are defined. The primary path corresponds to the physical link while the backup path is Layer-1-disjoint from the physical link and when the link fails, the pseudo-wire can be rapidly restored. After a single link failure occurs and the primary path fails, the traffic is rapidly switched over to the pre-calculated backup path. Thus, single link failures are transparent to the Interior Gateway

## IEEE COMSOC MMTC E-Letter

Protocol (IGP). Higher-layer packet loss recovery methods (such as retransmissions and Forward Error Correction (FEC)) can handle most losses of 50 milliseconds or less without significant viewer impairment. But, FRR must be coordinated carefully with the placement of network capacity to avoid traffic overlap during network failures, which can cause congestion and packet loss. For real-time video services, loss from even mild congestion often has the same effect on the customer's perception of service as an unprotected link failure. Therefore, we avoid congestion due to traffic overlap upon a single link failure by constructing the multicast tree carefully to be non-overlapping from each link's backup path [8]. However, multiple concurrent failures may still cause traffic overlap with FRR. It can take a fairly long time to identify, isolate and repair a link failure in the IPTV backbone. For instance, it can take up to 3-4 hours to repair a backbone link failure. Thus, multiple concurrent link failures (an additional failure occurring during the time it takes to repair the first) are not necessarily rare. To overcome this, we have developed a cross-layer restoration approach that combines both FRR-based restoration for single link failure and "hitless" (i.e., without loss) PIM-SSM tree reconfiguration algorithms to prevent traffic overlap when multiple failures occur [4]. The proposed methodology makes the IGP layer aware of the failure, notifies the multicast agent of the routing changes, re-converges the network quickly to the new multicast tree, and takes down the FRR backup path. The protocol is constructed carefully so that the FRR mechanism is used only for a short period thus allowing the multicast tree to dynamically adapt to any additional failures that might occur during the first link outage, all in a "hitless" manner. Thus, our proposed approach makes the IPTV backbone infrastructure much more robust to failures, including multiple concurrent link failures. More details may be found in [4].

### **Multicast-based Fast Channel Change for IPTV**

As mentioned before, IPTV places significant bandwidth demands on the infrastructure. There are both steady state and transient traffic demands. One of the sources of transient bandwidth demand is from users switching channels. The transient traffic demand can be significant in terms of network bandwidth and video server I/O capacity, to meet the user's

quality of experience goals. To understand this, let us examine how IPTV is delivered. Using IP multicast, a distinct multicast group is utilized for each TV channel. The consumer's set-top box "tunes" to a particular TV channel by joining the multicast group for that channel. There is a channel switching latency due to the time taken to join the multicast group and (more significantly) a time to wait for an I-frame and then fill up the play-out buffer for the new channel (to prevent underflow and stalling the playout). The total channel switching latency can be several seconds, causing unacceptable quality of experience to users who may want to surf or browse channels. In contrast, the channel change time with alternate distribution methods is minimal. To minimize this delay, a technique generally called "Instant Channel Change" (ICC) (there are several other names used in the industry for similar mechanisms) has been proposed. One approach first sends a full-quality unicast stream of the new channel's data at a higher bit rate [9]. The higher bit rate allows the play-out buffer to be filled faster and masks the time to join the different multicast group for the new channel. But, each such unicast stream adds a transient demand. Temporally clustered channel change requests (e.g., during prime-time viewing where subscribers switch at the completion of a program), from users can impose a substantial demand on network and server resources. In particular, it poses scalability challenges for the metro and access networks and for the server infrastructure at each VHO.

We have developed an approach where the bandwidth requirements are relatively independent of the population of users switching to a channel or the arrival process of the channel change events, by staying away from resorting to the use of a unicast stream when a user changes a channel. Our multicast-based approach using a secondary "channel change stream" associated with each channel. This secondary stream is a lower bit rate stream carrying only I-frames and associated audio. During a channel change event, the client does a multicast join to this secondary stream and results in a low "channel display latency". Concurrently, in the background the play-out buffer of the new full-quality multicast stream is filled. Once the playout point is reached, the full-quality video is displayed and the transition to the new channel is complete. The time interval from when the user requests the channel change to this completion point we call as the "channel switching latency".

## IEEE COMSOC MMTC E-Letter

We expect that the secondary channel change stream carrying I-frames is likely to use about 50% of the bandwidth compared to the full-quality video. The drawback therefore is the 50% additional capacity required during the channel switching period on the access network to the user, which we believe is manageable. Our proposed scheme also requires additional I/O capacity from the video server (up to 50% for each channel that has at least one channel change in progress). However, unlike the unicast ICC which requires a full-rate stream at an accelerated rate for every single user (of which there may be thousands if there is a correlated channel change event), the multicast approach uses a single lower rate I-frame stream and is relatively independent of the user population requesting a channel change. Moreover, we have shown through traces that the maximum simultaneous number of channels experiencing a channel change event at any time in an operational environment is small compared to the total number of active channels.

Our multicast based channel change approach has several performance benefits including lower bandwidth consumption during a burst (flash crowds) of channel changes (the obvious advantage of using multicast), lower channel display latency (50% lower), and lower variability of network & server load. For a low intensity of channel change events, the reduction in the display latency with the multicast approach may not be significant. But, even with a modest rate of channel changes, the need to use separate unicast assist sessions leads to consistently higher bandwidth requirements than our multicast scheme. The predictable load on the links and video servers with the multicast scheme makes provisioning decisions much simpler. The tradeoff in our proposed multicast scheme is the use of a lower quality (just I-frames) video displayed during the play-out buffering period. We believe that this is a tradeoff worth making to deliver the quick response to clients and the benefit of reduced bandwidth requirements on the metro and access network and the VHO server infrastructure. More details may be found in [5].

### **Adaptive Video-on-Demand with Cooperative Peer Assists and Multicast**

A discussion of IPTV wouldn't be complete without looking at the oncoming challenge of having to deliver a larger and larger share of entertainment to users "on-demand". We

envision a future in which almost all content including television shows, user-generated videos, and full-length movies will be viewed on-demand. The IPTV infrastructure will be used for delivering the video-on-demand (VoD) content to homes, in addition to streaming live (linear-TV) content. An IPTV provider who may also own the network infrastructure has the option to choose from or combine different data delivery mechanisms for delivering on-demand content. Initially, as the service of on-demand content is in its early stages, streams could just be unicast to each client from a server. But again, as with the unicast channel change solutions, this presents scalability challenges for the metro and access networks and for the server infrastructure at each VHO. We could seek to use a multicast based approach for serving content on-demand, as has been suggested in approaches proposed over the last 15 years or so for near-on-demand VoD. But, this requires clients that are willing to tolerate a (configurable) delay for the start-up of the video. Another viable alternative is to use peer-to-peer approaches with entire streams cached on "seed" nodes, and then "swarmed" onto requesting clients using peer-to-peer (P2P) transfer. While this is very attractive for popular content, it may be difficult to use to deliver less popular content to users. Thus, each approach has its advantages, and possible drawbacks. Ideally, the service provider would like to deploy a single mechanism that automatically adapts to the prevailing conditions and still meet its goals of providing a robust, scalable and flexible service that can accommodate a large subscriber base and an equally large content library. Our work on using Cooperative Peer Assists and Multicast (CPM) dynamically combines multicast, unicast, peer-transfer and pre-population in a seamless manner to meet this requirement.

CPM approaches the problem of content distribution from a service provider's perspective, by using trusted servers that maintain a master copy of each piece of content, and cooperative peers that assist in sharing and transferring content that they have cached. When a piece of content is requested, CPM identifies the best source of the content. This decision is made dynamically and depends on the cost of transferring the content from that source. A key aspect in CPM is that content is broken up and retrieved in relatively small chunks (similar in nature to a number of recent approaches). Each chunk is typically 30 seconds in length and is identified using a unique identifier. These



## IEEE COMSOC MMTC E-Letter

chunks serve as points where a new download decision can be made. In CPM, a client can switch from peer transfer to a server transfer and vice-versa for each new requested chunk depending on whichever results in lower transfer cost. The client dynamically determines and requests each chunk from the most appropriate source of the chunk. In practice, a client obtains a small number of chunks in parallel. Depending on the amount of time available, which we call 'slack', the client may choose to directly go to the server or try a peer transfer. The strength of CPM is its ability to adapt to a wide range of situations. CPM combines the effectiveness of server multicast at high request arrival rates with the power of peer-to-peer download that excels at low to moderate arrival rates, to provide uninterrupted playout of requested content in a manner that is relatively independent of the popularity of the content.

### Summary

Having a robust and scalable infrastructure for delivering video over an IP network is a key enabler for introducing additional capabilities to enhance the user's viewing experience. We envisage delivery over IP networks will provide the capability for users to interactively obtain "what they want, when they want, where they want" in terms of high quality entertainment and information. Our work is aimed at providing this basic foundation for such an IP infrastructure.

### Acknowledgements

I thank the large number of colleagues and collaborators who have worked with me on several of these issues related to IPTV, including Robert Doverspike, Vijay Gopalakrishnan, Rittwik Jana, Guangzhi Li, Kostas Oikonomou, Rakesh Sinha, Divesh Srivastava, Dongmei Wang, Chris Chase, Damodar Banodkar, Bobby Bhattacharjee and Shivkumar Kalyanaraman. Much of the work discussed here is due to their hard work and creativity.

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## **IEEE COMSOC MMTC E-Letter**

**K. K. Ramakrishnan** is a Distinguished Member of Technical Staff at AT&T Labs Research in Florham Park, New Jersey. He joined AT&T Bell Labs in 1994 and has been with AT&T Labs. Research since its inception in 1996. Between 2000 and 2002, he was at TeraOptic Networks, Inc., as Founder and Vice President. Prior to 1994, he was a Consulting Engineer and Technical Director in Networking at Digital Equipment Corporation.

His current research interests are in networking and communications, including congestion control, multimedia distribution, content dissemination and problems associated with large scale distributed systems. He has published over 100 papers and has nearly 90 patents issued. He is an IEEE Fellow and an AT&T Fellow, recognized for his work on

congestion control and VPN services. His contributions on congestion control, channel access protocols (for Ethernet and Cable), network interfaces, operating system support for network I/O, VPN services and IP Telephony have been adopted and implemented in the industry.

K.K. has an MS degree from the Indian Institute of Science (1978), an MS (1981) and Ph.D. (1983) in Computer Science from University of Maryland, College Park. K.K. has been on the editorial board of the IEEE/ACM Transactions on Networking and IEEE Network Magazine and has been a member of the National Research Council Panel on Information Technology for NIST. He has participated in numerous standards bodies working on communication networks.

## Network-Reconvergence Technologies for Low-loss Video Transport

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### 1. Introduction

The Internet Protocol (IP) is becoming the dominant network technology for video transport and in turn, video is becoming an increasingly significant component of IP network traffic. In the January and March editions of IEEE Internet Computing Magazine, we described the IP service-level requirements for a transported video service and considered the network factors that impacted the viewers' quality of experience (QoE) for IP-based video streaming services such as IPTV and Cable TV (1) (2). We also highlighted the impact that different durations of IP packet loss had on the viewers' QoE, presenting results showing that due to the temporal compression employed in video encoding, even short durations of IP packet loss could result in significant visual impairments, e.g., even a single IP packet loss may result in several hundreds of milliseconds of visible impact to the viewers.

Considering the primary causes of packet loss in IP networks – congestion, lower-layer bit errors and network element failures – whilst the Differentiated Services (RFC 2475) and Integrated Services Architectures (RFC 1633) may be used to engineer a core IP or Multi-Protocol Label Switching (MPLS) network to help deliver the required delay, jitter and loss rates for a video transport service (3), packet loss can still occur due to network reconvergence events (link or node failures or recovery) or lower-layer errors, which might result in visual impairments to an impacted video service.

There are a number of IP / MPLS network technology approaches to minimize the duration of packet loss experienced due to network reconvergence events, such as Fast (IP routing protocol) Convergence, MPLS Traffic Engineering (TE) Fast ReRoute (FRR) and Multicast only Fast ReRoute (MoFRR), but packet loss may still be experienced, which will impact the perceived QoE. These network-level technologies may be further augmented by application or transport-level approaches to recover from any loss experienced due to either reconvergence events or lower-layer errors, and hence, improve the viewers' QoE. Such approaches include Forward Error Correction (FEC), retransmission, temporal redundancy (i.e., sending the same source data twice separated in

time) and spatial redundancy (i.e., using redundant video streams that use paths that are spatially diverse at the network level).

In this article, we give an overview of the different network-reconvergence technologies and briefly compare a number of different deployment models employing these capabilities that can be used to minimize and manage video packet loss, and hence, potentially improve the quality of a broadcast IP video service, where a source is originating broadcast video content, which is distributed across core and aggregation networks, which may be IP or MPLS, to the last-mile access connection and the subscribers.

The combinations of different network-reconvergence technologies and loss-recovery approaches may result in a potentially confusing number of possible deployment models, which have different pros versus cons and cost versus complexity trade-offs. In a future article, we are planning to examine these pros, cons and trade-offs in more detail in the context of lossless video transport.

### 2. Comparing Approaches

We compare different deployment models in terms of the following characteristics:

- *QoE*: In comparing the different approaches for video transport, we need to consider whether the models are lossy or lossless. For lossy approaches, we compare the video impairments resulting from the anticipated packet loss. We define a lossless video transport service as one where the resulting QoE of the video viewer is not impacted by loss caused by network reconvergence events or lower-layer errors.
- *Capacity Provisioned and Consumed*: It is a standard practice for IP / MPLS transport networks to be built with redundancy, in all but the last-mile access connection to the subscriber, such that – where possible – they are resilient to single element (link or node) failure cases. It is also normal for service providers to provision sufficient capacity on both working and failure-

## IEEE COMSOC MMTC E-Letter

case paths to support their premium services. When comparing the different approaches, we consider the link bandwidth consumed (and hence how much needs to be provisioned) per video stream on both working and failure-case paths, relative to the basic video stream bandwidth.

- *Delay*: Different models have different impacts on the delay of the transported video stream. Delay introduced by some models may have an adverse impact on the interactivity experienced by the viewers.
- *Deployment Considerations*: These include requirements imposed on the network in order to support the approach and the overall design, deployment and operational complexity of network in support of that model.
- *Applicability*: These factors will impact the applicability of the different deployment models. Not all models will be suitable everywhere; is it viable for the core network and for the access and aggregation?

An idealized video transport solution would be lossless in the presence of packet loss due to network failure cases and lower-layer errors, would be bandwidth efficient, requiring that only the basic video stream be provisioned on working case and failure case, would add negligible overall delay to the transported video stream, and would not significantly increase core network cost and complexity with respect to design, deployment and operations. Such a solution may apply different models in different parts of the network.

### 3. Network-Reconvergence Technologies

Networks are built resiliently, so that on the failure of any single component (e.g., link or router) between the source and the last-mile to the subscriber, an alternate path will be available. The last mile to the subscriber is normally a single point of failure due to cost constraints. In the rest of the network, however, network element failures may cause packets to be dropped until connectivity is restored around the failed network element. The resulting loss period depends upon the underlying network

technologies that are used; there are a number of potential choices that can result in loss periods from tens of milliseconds upwards. In this article, we consider three different approaches:

- Fast (IP routing protocol) Convergence
- MPLS Traffic Engineering Fast ReRoute (MPLS TE FRR)
- Multicast only Fast ReRoute (MoFRR)

#### *Fast (IP Routing Protocol) Convergence*

Reachability or connectivity in IP networks is established by Interior Gateway (routing) Protocols (IGPs). For IP unicast traffic, after a network element failure, even if there is an alternate path around the failure, there will be a loss of connectivity that causes packet loss until the routing protocol establishes an alternate path. The process of finding alternate paths is called *convergence*, and *convergence time* is the time between an event occurring, which causes a routing protocol recalculation (such as a link or node failure or recovery), and convergence being achieved. The convergence time is comprised of time to detect the failure, time to propagate the news of the failure through the network, and time to recalculate new paths at nodes or routers whose forwarding is impacted by the failure. The process of network reconvergence may need to complete at multiple routers before end-to-end connectivity is re-established.

Multicast provides the capability to send traffic to a group of interested receivers. This is achieved within IP networks through the creation of multicast distribution trees (MDTs) that connect the source, which is the root of the tree, to the group of interested receivers via a number of branches and replication points. IP multicast is the default technology used for IP-based broadcast video services, such as IPTV and Cable TV, because of the bandwidth efficiencies it offers for point-to-multipoint services; the source needs to send a packet only once, even if it needs to be delivered to a large number of receivers. IP multicast connectivity is dependent upon both multicast routing protocols and unicast routing protocols. Protocol Independent Multicast Source-Specific Multicast (PIM-SSM) (RFC 4607) is commonly used as the multicast IP routing protocol for point-to-multipoint streaming video services. With PIM-SSM, the network builds a separate MDT rooted at each multicast source and clients connected to the tree

## IEEE COMSOC MMTC E-Letter

receive content directly from the source<sup>1</sup>. The MDT is built from the receivers (i.e., the leaves) to the source (i.e., the root), and in doing so PIM-SSM relies upon the unicast routing tables established by the IGP to determine which is the preferred route, and hence, the interface towards the source; multicast traffic on a particular MDT is only accepted if it is received on the preferred interface towards the source. Following an event such as a link or node failure or recovery the preferred path, and hence, the interface towards the source may change, and therefore the multicast convergence time is dependent on the unicast convergence time.

We consider a network deployment where a broadcast video service utilizes a source in a headend that is originating a single stream per broadcast channel that is distributed across the core, aggregation and access networks to receivers using standard IP multicast, with fast convergence used to reduce the loss of connectivity resulting from reconvergence events such as core link or node failures. Such a deployment has the following characteristics:

- *QoE*: On the failure of a core link or node, there will be a loss in connectivity for that stream until the IGP reconverges on the alternate path; in well-designed networks, where the routing protocol implementation is optimized for fast convergence, for 400 multicast groups (potentially equivalent to 400 broadcast TV channels), convergence times of approximately 300 ms are realistically achievable today. Considering the results we presented in a previous article (2), the resulting video impairments on an impacted video stream may be visible to viewers for up to one second.
- *Capacity Provisioned and Consumed*: Service providers generally provision sufficient core network bandwidth to be able to support the IPTV load both in working-case and in single-element failure-case conditions. For this model, if  $B$  denotes the basic video stream bandwidth per channel, bandwidth  $B$  needs to be provisioned per channel on

working and failure-case paths. In normal working-case conditions, bandwidth  $B$  is only consumed on the working-case paths. Where the receivers are spread around the topology, however, as is commonly the case with IPTV, the net effect is that the working-case paths for some receivers are the failure-case paths for other receivers, hence, in practice bandwidth  $B$  will be consumed on both working and failure-case paths most of the time.

- *Delay*: This model adds no additional delay to the transported stream, i.e., the delay incurred is just the time it takes for the packets to be transported across the network.
- *Deployment Considerations*: This approach is the simplest in terms of network complexity in that no extra requirements are placed on the network.
- *Applicability*: This model may be applied in both the core and aggregation networks, and can be considered as a baseline for broadcast video deployments, against which other models may be compared.

### *MPLS Traffic Engineering (TE) Fast ReRoute (FRR)*

MPLS TE (RFC 3209) uses the Resource ReSerVation Protocol (RSVP) (RFC 2205) to signal MPLS TE tunnels, which can be used to explicitly define which paths through the network that video connections take and provides admission control and bandwidth reservation along the paths. Enhancements to MPLS TE supporting point-to-multipoint services have been defined in RFC 4875. Where MPLS TE is deployed, FRR (RFC 4090) can be used in addition to fast convergence. MPLS TE FRR is a local protection scheme, which enables pre-calculated backup TE tunnels to be used to protect against link, interface and linecard failures. On occurrence of a failure, there are no delays associated with the distribution of updated routing information or due to routing table recalculation prior to connectivity being restored, unlike IGP convergence which is a distributed computation process. Consequently, where an alternate path exists, restoration of connectivity following network element failures is always

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<sup>1</sup> This may be contrasted with Any Source Multicast (ASM) approaches, such as conventional PIM-Sparse Mode (SM) (RFC 4601) where clients may receive content via a Rendezvous Point.

## IEEE COMSOC MMTC E-Letter

likely to be faster with MPLS TE FRR – typically within 50 ms – and more deterministic than those achieved with IGP fast convergence.

We consider a network deployment, as per the previous model, where a source in a headend is originating a single stream per broadcast channel, which is distributed across the core, aggregation and access networks, however, unlike the previous model, MPLS TE FRR is used to reduce the loss of connectivity following core link or node failure cases. Such a deployment has the following characteristics:

- *QoE:* Where MPLS TE FRR is used, the loss of connectivity following core network failures is lower (typically less than 50 ms) and more deterministic than with IGP fast convergence, and is independent of the number of multicast groups. The results in (2), however, demonstrate that the resulting video impairments may still be visible to viewers for up to ~700 ms.
- *Capacity Provisioned and Consumed:* In working-case conditions, the bandwidth consumed for this model is the same as for the previous model. Where point-to-point MPLS TE FRR backup tunnels are used to protect multicast services, however, there may be traffic duplication (with link protection) or worse (with node protection) whilst the MPLS TE FRR protection is active following network element failure cases, i.e., before the network has converged around the failure. This issue is only present as a transient, which lasts from the time of the failure event until the network has converged, however, this is the time period over which the deployment of FRR may be expected to provide better QoE for an IPTV service than fast convergence. Therefore bandwidth  $\geq 2xB$  needs to be provisioned on failure-case paths in order for the faster rerouting capabilities of MPLS TE FRR to be effective.
- *Delay:* This model adds no additional delay to the transported stream, i.e., the delay incurred is just the time it takes for the packets to be transported across the network.

- *Deployment Considerations:* This approach implicitly requires that MPLS TE is deployed, which incurs an additional level of complexity for network design, deployment and operations. Further, additional planning is required to make sure that backup MPLS TE tunnels are diverse from the primary elements they are protecting.
- *Applicability:* The bandwidth requirements of using MPLS TE with FRR may limit its applicability to core networks, where  $2xB$  overprovisioning of video capacity may be acceptable for transient periods.

### *Multicast only Fast ReRoute (MoFRR)*

MoFRR (4) is a simple enhancement to PIM-SM processing that can provide the capability to instantiate multiple branches of the same multicast tree between a source and receiver. If the branches are spatially redundant, MoFRR can be used to further reduce the effective multicast convergence time, as when a network link or node failure impacts one branch, the device implementing MoFRR simply needs to detect the failure and switch to the working branch.

We consider a network deployment, where a source is originating two resilient multicast streams carrying the same content. A router between the source and the last-mile access connection will receive both streams, from which only one will be forwarded on. MoFRR is used on that router to instantiate the resilient multicast streams. Such a deployment has the following characteristics:

- *QoE:* MoFRR is effective at reducing the losses experienced due to network element failures relative to the IGP fast convergence. MoFRR can potentially reduce effective times to sub 100 ms (5); correlating with the results in (2), we may conclude that the resulting impairments may still be visible to viewers for up to ~750 ms.
- *Capacity Provisioned and Consumed:* There is a common misconception that approaches that use resilient multicast streams use additional network capacity compared to models which use a single stream or which use FEC. Where the receivers are spread around the topology and there are multiple paths

## IEEE COMSOC MMTC E-Letter

back to the source, however, as is commonly the case with IPTV, the net effect is that the working-case paths for some receivers are the failure-case paths for other receivers. Hence, the resilient multicast streams are generally already established. Therefore in practice, if  $B$  denotes the basic video stream bandwidth per channel, bandwidth  $B$  needs to be provisioned per channel on working and failure-case paths, i.e., the same as for the IGP fast convergence case.

- *Delay*: This model adds no additional delay to the transported stream, i.e., the delay incurred is just the time it takes for the packets to be transported across the network.
- *Deployment Considerations*: This approach is relatively simple to design and operate; the only network requirement is for MoFRR, which is simple to design, deploy and operate; if configuration is required, it is a simple configuration.
- *Applicability*: This model may be applied in both the core and aggregation networks.

### 4. Summary

The following conclusions are drawn from the preceding analysis and the related studies:

- All approaches may experience visible impairments resulting from losses due to network reconvergence events. These impairments are comparable for MPLS TE FRR and MoFRR (~700 ms and ~750 ms, respectively), and comparably lower than for IGP fast convergence (~1 s).
- IGP fast convergence and MoFRR require that the least bandwidth ( $B$ ) is provisioned on working and failure-case paths, whereas MPLS TE FRR requires that  $2 \times B$  bandwidth is provisioned on failure-case paths.
- IGP fast convergence and MoFRR are simpler solutions to design, deploy and operate than MPLS TE FRR.

Where the packet loss caused by reconvergence events or lower-level errors is too high, these network-level technologies may be further augmented by application or transport-level approaches to recover from any loss experienced. In a future article, we are planning to examine the relative benefits of a number of deployment models, which combine different network-reconvergence technologies and loss-recovery approaches.

### Acknowledgments

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## IEEE COMSOC MMTC E-Letter



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## KreaTV in focus: The settop box and beyond

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**Abstract**—TV has looked more or less the same to the end consumer for a long time, but at Motorola we see a change in attitude among our customers. They are starting to use the power of the tools they have at hand. The set top boxes are not just used for watching TV anymore.

**Index Terms**—Motorola, IPTV, TV, settops, KreaTV, STB

### 1. TRADITIONAL IPTV

Up until recently our KreaTV based set top boxes have been deployed with a headend which multicasts normal TV channels with program guides and possibly provides a video on demand service. This is done in a traditional geographically well-defined network with a headend, backbone and an end user who has some kind of last mile connection. Depending on the last mile speed the end user may or may not be able to watch High Definition TV.

This has been a cost effective method for distribution of TV in areas where an existing network is already in place. A video on demand service can easily be added into these environments.

In other words, the set top box enables end users to watch TV or rent a movie, just like they already have been doing for quite some time now.

### 2. TRENDS

During the last two years, we at Motorola have seen a shift from the traditional IPTV setup to more advanced configurations which offer the end users richer functionality. One example is the KDDI au BOX project where Motorola was presented with a concept that was not even TV centric. KDDI wanted to sell the box as an accessory for their mobile phone. A whole new set of features were implemented, among them transcoding and mobile phone synchronization. In November 2008 KDDI began selling a solution where the video service was completely on demand, with the capability to transcode and redirect the video to the mobile phone. The end users could also buy music, play DVDs, and use home networking among other features [1].

We have also seen special soccer channels that show several games and live betting. The video footage from the main game occupies most of the screen, but there are also smaller embedded windows which show other games. If something interesting happens you can change audio track to one of the other games.

Some of our customers are deploying over-the-top set top boxes that connect directly to the Internet. In this way they can expand beyond the confines of their local network and reach end users they had no chance of reaching before.

There are also customers implementing their own custom video on demand features. One example allows the importing of DVDs into a centralized media library and then reproduces the typical DVD menus for the end users, so that the experience is the same as if they had bought the disc themselves.

All of this is performed using existing technology that has been around but that was not intended for these exact applications, and they give the end user something more than just a TV feed or a video on demand service.

### 3. KREATV

The enabling software in all of the deployments discussed above is KreaTV. Built on Linux, it provides a foundation that enables the extension of most aspects of the software.

To start at the beginning; the KreaTV software is running on a settop box [2]. This box could have several different hardware setups and it does not even have to be a Motorola box. The demand KreaTV places on this box is that it implements a well defined Hardware Abstraction Layer. All our Hardware Abstraction Layers build on Linux kernels that are more or less adapted by the chip vendor and take care of primitive function calls down to the hardware. The Linux kernel is of course extendable with kernel modules allowing new hardware to be integrated. The natural point of extension that is present on most of our hardware platforms is the USB port.

## IEEE COMSOC MMTC E-Letter

On top of the Hardware Abstraction Layer we have an Application Platform which simplifies application development. The Application Platform takes the abstraction of the box hardware to yet another level as it handles calls that do not have a corresponding hardware feature. It also has functions that are more advanced than the Hardware Abstraction Layer, taking some of the logic out of the applications. This means that applications developed for the KreaTV platform are independent of whatever hardware they are running on.

The platform also offers a method to extend or replace the functionality of the decoding software. A format or transmission method not currently supported by the box can be implemented here. The best examples of this type of extension are the decryption systems that are added by all the conditional access vendors we support. A good example of replacing functionality is the DVD-menu video on demand service mentioned earlier.

The easiest and most popular way to produce graphical user interfaces, and to control the box hardware, is to use the combination of JavaScript and HTML. Doing so can allow rapid application development and proof-of-concept. It is done via an extended web browser that forwards API calls down to the Application Platform. The browser can also be extended further by developing plugins which allow faster execution of selected functions, or to add extra functionality. This is described in figure 1.

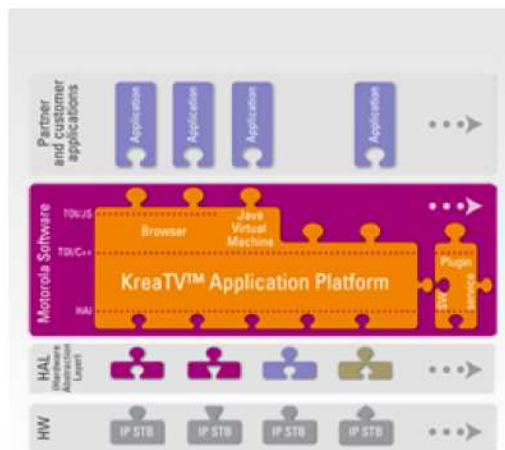


Figure 1. The KreaTV platform

Most development on the box is done in C/C++, but the TV Open Interface (TOI) can also be accessed through a variety of other languages like Java, shellscripts and the above mentioned JavaScript. If there is a need for another language it can actually be implemented with an IDL compiler for the interface, a cross compiler to compile the code into the executable for the architecture of the box, and some code for the actual inter process communication. [3], [4]

#### 4. NEAR FUTURE POSSIBILITIES

As we have seen, KreaTV customers have no real limitations regarding the extension of the software. The easiest way to start combining technologies is probably to create portal pages that give access to various web sites. This offers an experience designed directly for display on a TV instead of a computer screen. We have seen aggregation sites on the Internet, so why not also in the TV environment?

We can also integrate video into these web pages, so there are possibilities for more advanced program guides, video shows and so on.

From the service providers' point of view, reducing channel change times is very desirable. The set top box might be pushed into masking these channel change times with a system similar to the following. Where simultaneous decoding of several channels is available, channel change times can be almost eliminated by always remaining tuned into a special mosaic channel. This mosaic channel, which contains a grid of all the other available channels in scaled down versions, is always tuned to in the background. When the end user requests a channel change, the mosaic channel is shown on screen, but zoomed in on the desired channel so it is the only thing visible. Naturally, the zoomed in picture will not be of the same quality as the original one, but this effectively masks the buffer and join times while the real channel is being tuned.

For even more enthusiastic developers, CPU intense algorithms can be implemented in C++ and be accessed via the plugin functionality of

## IEEE COMSOC MMTC E-Letter

the browser to take advantage of the faster execution. If the faster execution is desirable throughout the whole application then everything can be implemented natively.

Native code can also access the underlying Linux environment, including kernel module products like proc entries and device nodes. As there are means to build kernel modules with the KreaTV Software Development Kit, new hardware can be added and presented to the upper layers of software. This means that the boxes are not only configurable and extendable in software, but also in hardware.

I would like to see the USB port being used as an extension point. It could be very interesting to add a simple hardware interface for electronics- and do it yourself-hobbyists to explore. The hardware interface could then be propagated through a Linux kernel module, making its way to the JavaScript layer via a browser plugin. Users could then submit their own JavaScript to allow control over their own little gadgets. Service providers could support communities where ideas are exchanged and gadgets bought and sold.

An excellent example of this can be seen on Erik Freiholtz web page [5], where temperature sensors can be purchased. The temperature sensor is connected to the computer and the readings submitted to the web server. When the server has collected a reading it presents it on the web page. There is also the possibility to download hardware schematics and software to produce your own temperature sensor.

I think introducing environmental sensors in the TV environment would have some value, but there are a lot of other things that can also be implemented through this same interface. Some examples include gaming controls, home

automation, voting mechanisms and useless but fun gadgets.

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**Is IPTV a quality experience? Quality monitoring in an IPTV network**

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When delivering video services, it is essential to deliver a high-quality video signal with interesting content. While it is well known that viewers rate compelling content as having a higher perceptual quality than boring content [1], I won't discuss here how to solve the problem of guaranteeing that the content is interesting to the viewers [2]. Instead, I'll discuss technological aspects of determining when the video signal itself has high quality. While audio is clearly important, I'll assume here that it is not.

Video quality measurement is necessary in an IPTV system prior to initially deploying the system, for adequate provisioning and to select the best encoders, set-top boxes, and other components. It is also necessary when establishing a new customer to ensure adequate initial quality and to provide a benchmark to compare quality at a later point. Once the system is up and running, there is a need to monitor the video quality on an ongoing, operational basis.

Good quality video only appears on the viewer's TV when everything goes right. Impairments can be introduced either by the content provider, the service provider, the network provider, or on the customer premises. Hence, end-to-end quality assessment is important so the operator knows the quality of video being presented to the viewers. This requires measuring at various points along the processing chain. The goal of a well-designed video monitoring system is to detect and report problems in the distribution chain so that the problems can be fixed before any customers call to complain.

A variety of faults can occur. A packet loss can cause annoying degradation of the video signal. Insufficient bit-rate can cause a video encoder to output video with heavy and annoying compression artifacts. Poor content acquisition can cause aliasing artifacts, for example, if format conversion from HD to SD (or vice versa) is not done correctly. Lip synchronization may be lost, often in the set top box itself. Viewers can tolerate most of these, provided the content being sent is interpretable in some way or another by the human eye. The worst possible failure is a black screen.

There are three essential ingredients for an effective video monitoring system. First, one must be able to insert a probe to access useful data at the monitoring point. Second, one needs

accurate algorithms to process the extracted data. Third, one needs an efficient reporting structure to get the processed data to a central location for comparing results and generating alarms. Clearly, one without the other is ineffectual, since fancy reports of numbers that have little bearing on actual quality will not help solve anything!

Video quality assessment algorithms are frequently categorized by the type of information they use. Full-Reference (FR) metrics use the original pixels and the decoded pixels. Reduced-Reference (RR) algorithms extract key information from the original and decoded pixels; this information is then sent to a common location where the two sets can be compared. No-Reference algorithms may be based on decoded pixels (NR-P), the lossy bitstream (NR-B), or both (NR-BP). The latter is being standardized under the poorly-descriptive term "hybrid metrics" in ITU SG-12 [3].

In-network monitoring of video quality imposes structural and processing constraints. FR algorithms are not feasible inside the network, because the original video is rarely available inside the network. However, they can be very effective to provision the network or benchmark encoders. RR methods may have a role in network-based monitoring if a reliable side-channel is available for the information extracted from the original pixels to be transmitted to the quality monitor. For example, a blackout detector like the Video Outage Detection Algorithm [4] could be run at the content generation site and the occurrence of known, desired blackouts in the video content could be sent to the monitoring point. Then when VODA is applied at various points in the chain, alarms can be avoided when known blackouts are encountered. Such information, whether in-band or out-of-band, would need to be standardized so that all pieces of the distribution chain can effectively work in tandem.

NR-P methods require a decoder for every stream in the network, so they may still not have general applicability at all points in the chain. Further, they may not be viable for proprietary compression algorithms or encrypted video payloads. NR-B methods do not require a complete decoder for every stream [5]. Full-Parse NR-B methods use variable length

## IEEE COMSOC MMTC E-Letter

decoding to obtain source and compression parameters. This allows algorithms to improve their performance for the specific source content. QuickParse NR-B methods require the least processing among the options described, but are still capable of providing valuable information regarding networked video quality [6].

Many open problems remain in this area, and there is a strong desire from IPTV providers to obtain viable solutions that correlate well with human assessment.

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## Serving Personalized Content in IPTV Platforms

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### I. Introduction

Internet Protocol TV (IPTV) is a system to deliver television content to customers over IP-based network. Different from the classic cable TV services, IPTV platforms provide customers with tremendous amount of multimedia content with rich interaction and personalized services. Serving personalized content, including the targeted advertisement, favorite live and on-demand content, and so on, is essential for IPTV service providers to create new revenue streams, attract more customers, and improve customer retention in a highly competitive market. In this letter, we discuss the opportunities and challenges of serving personalized content in IPTV platforms.

### II. Content Personalization

While IPTV services provide hundreds of channels of TV programs, thousands of titles of video on demand content, and millions of user generated content (for example, YouTube video), the amount of content served in IPTV system far surpasses the consumption capability of regular consumers. A challenging task is to locate the relevant content (either pre-recorded, live, or scheduled) for the users based on their interests. Personalized service is a promising solution for this challenge and IPTV systems have all required infrastructure to support such a value added service.

Liu et al [1] created a content personalization and adaptation prototype for three-screen services, including IPTV system. Video content is ingested from broadcast services continuously or alternatively in a mode where selected serial content is captured as a set of episodes. Metadata and semantically meaningful index of the content are extracted to facilitate query response. The main content analysis components include shot boundary determination, face detection and tracking, automatic speech recognition, closed caption alignment and normalization, named entity extraction, speaker segmentation, anchorperson detection, and multimodal story segmentation. The goal is to logically partition a long-form video (more than 30 minutes) into short and coherent units (1 – 10 minutes) that focus on certain topics. In [1], the users define their topics of interest in their profiles by choosing a set of keywords and

key phrases and specifying the preferred program categories, including program sources and genres. The content personalization engine identifies relevant content from the collections by applying the user profiles and generates descriptions of content sets for each user. Users then can access their favorite content that is assembled automatically. Powered by such personalized service, the users no longer need to actively search the content on IPTV or waste time on browsing irrelevant materials, but focus on the personalized content prepared by the system. The interface for set-top browsing with a remote control including extracted thumbnails and processed metadata is shown in Figure 1. In this example, the user has defined four topics: Science, Telco News, My Teams, and Politics. We can imagine that the topic, My Teams, could be defined by the following keywords: baseball, basketball, hockey, NBA, NHL, MLB, etc.



Figure 1. An IPTV interface for browsing personalized content.

Another application of personalized service is to provide addressable advertisements to the end users. With the knowledge of the customers' niche and desire, the targeted commercials are more enjoyable. For example, if a customer is planning to buy a new car, automobile advertisements are welcome and desired. In such case, the value of the addressable advertisement also increases. The targeted advertisement insertion definitely breaks the traditional advertisement model where non-targeted commercials are treated as intrusions and are preferred to be skipped in most cases.

To provide personalized service, the system needs to be aware of the identifications of the current

## IEEE COMSOC MMTC E-Letter

viewers in front of the TV. Different viewer identification and authentication techniques can be applied to IPTV system. These include biometric person identification, for example speaker verification, face recognition, fingerprint; radio-frequency identification (RFID) tags; or the subscriber authentication schemes in the IP Multimedia Subsystem (IMS) framework; or a simple login session with user id and password.

While the greater interactivity and personalization are the advantage of IPTV systems, many technical challenges for delivering a seamless interactive and personalized IPTV experience still exist. For example, how to build an interface that allows the user to navigate all services intuitively and effectively? How to simplify the user interface such that non-tech savvy also feels comfortable? How to minimize the response time to meet the requirement of real-time IPTV applications, for example, multiple player on-line gaming? How to automatically discover the users' interests based on their viewing history or feedback on viewed content? How to recommend movie titles or TV programs that make sense to different users?

Bell and Koren [2] proposed a new method for recommending movie titles for customers, and their approach won the first Netflix progress prize. Lyu et al [3] studied many scenarios for personalized IPTV services, and proposed open APIs for the personalized EPG and the target advertisement services. Since 2006, Negev consortium [4] has been promoting research and development of personalized content services on TV. This consortium brings together experience and knowledge from both industrial and academic researchers to specify the next-generation infrastructure of personalized video services.

### III. Case Studies

IPTV services have been widely deployed in a number of countries. The leading IPTV markets include France, South Korea, Hong Kong, Japan, Italy, Spain, Belgium, China, Switzerland, Portugal, and United States. In this section, we use the AT&T's U-verse IPTV [5] service as an example to illustrate the state of the art of IPTV services available in the market.

In 2006, AT&T launched U-verse IPTV service, the first and the most advanced TV service in the United States. At the end of the first quarter of 2009, U-verse has more than 1.3 million subscribers. According to a J.D. Power and Associates study, U-verse TV ranked "Highest in Customer Satisfaction in the North Central, south, and West Regions." This is a proof of the

exceptional performance and reliability offered by the U-verse services.

U-verse packages normally bundles IPTV services with high speed internet access. Some captivating features of U-verse services include: 1) over 100 HD channels; 2) total home DVR; 3) Web and mobile remote access to DVR; 4) AT&T U Bar; 5) local Yellow Pages search; 6) AT&T Yahoo! Games; 7) Yahoo! Sports fantasy football; 8) advanced search functionality for programs and VoD library; 9) online photos from Flickr.

Among these innovative features, we want to emphasize a few that can be personalized. Figure 2 shows the AT&T U-bar, which allows the user to access customizable stock, weather, sports and traffic information on the TV screen, based on the preferences that the user set on the AT&T high speed internet portal.



Figure 2. AT&T U-bar.

In the Yahoo! Sports Fantasy Football, the viewers can track their fantasy team, including current team matchups and league standings, directly from the sports section of the AT&T U-bar. The customers can either join an existing league for free or create their own league.

With the seamless support of Flickr photos, the user can simply and conveniently watch slideshows and browse their online photos on the U-verse TV screen.

### IV. Challenges

In this section, we share our perspectives on the challenges of personalization services of IPTV systems, as a conclusion of this letter.

First – Content Management. IPTV provides virtually unlimited amount of content for the subscribers, and how to manage the content and provide personalized services for the end user in a scalable manner is still a big challenge.

## IEEE COMSOC MMTC E-Letter

Automating the personalization service and targeted advertisement insertion requires more content analysis technologies and user studies. The NIST sponsored TREC video retrieval evaluation (TRECVID) [6] is to promote progress in content-based analysis of and retrieval from digital video via open, metrics-based evaluation. The research efforts reported in TRECVID, including shot boundary determination, high-level feature extraction, content-based copy detection, surveillance event detection, video summarization, story segmentation, and video search, will surely enhance the content management in IPTV platforms.

Second – Mobility support. Customers do not want to be forced watching TV in front of the big TV screen in the living room. Mobility is definitely desired to enable IPTV services on the mobile devices, such that users can access the content anytime and anywhere. The IP Multimedia Subsystem (IMS), designed by the 3rd generation partnership project (3GPP) [7], is an open architectural framework for delivering multimedia content to a wide variety of mobile devices. Integration of IMS with IPTV definitely has potentials. How to manage the user authentication, right management, content adaptation and intelligent distribution, billing for IPTV services on mobile devices require more investigation.

Third – Interoperability. With more deployment of IPTV services from different telecommunication companies, interoperability will be an important issue. In the long run, open standards encourage the improvement of the IPTV technologies, and more importantly, benefit the IPTV customers. Many standard organizations, for example, ITU-T, ATIS, ETSI, DVB forum, Open IPTV forum [8], have devoted significant amount of effort to devising an open IPTV architecture. Yet, more attention is required to ensure the interoperability of the interactivity and personalization aspects of IPTV services.

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### 3D Games Deployment Platform for IPTV

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#### Abstract

Currently, web-based online gaming applications on IPTV are predominately utilising Adobe Flash or Java as their core technologies. These games are often casual, two-dimensional games and do not utilize the specialist graphics hardware which has proliferated across modern PCs and Consoles. Multi-user online game play in these titles is often either non-existent or extremely limited on IPTV platform. Computer games applications which grace the current generation of consoles and personal computers are designed to utilize the increasingly impressive hardware power at their disposal. However, these are commonly distributed using a physical medium or deployed through custom, proprietary networking mechanisms and rely upon platform-specific networking APIs to facilitate multi-users online game play. In order to unify the concepts of these disparate styles of gaming, we propose novel integrated development environments called Homura and NetHomura [12], and propose a framework for its deployment on IPTV. Homura is a game research project carried out by Liverpool John Moores University in collaboration with BBC R&D. Homura is available for download as open source in [12].

**Keywords:** IPTV, 3D Games, Java Web Start, Peer-to-Peer, Online Deployment

#### 1. Introduction

In 2004, the video game industry had annual revenue of approximately \$25.4B, and this has increased to \$54.6B in 2009. This figure represents a substantial 16.5 percent compound annual growth rate. Game platforms (both PCs and consoles) host non-networked and multiplayer networked games. Although revenue from offline games has been dominant over the years, analysts' predictions suggest that multiplayer online game sales will eventually dwarf those of traditional console PC games, with revenues approaching \$10.2B by 2010. Any industry which will join the game industry as a partner will take advantage of this growth.

The game industry is relatively young but had an important influence in the development of many

technologies in computing, communication and services; and for both hardware and software. While this development was for many decades diverging as shown in figure 1, the recent trend is that many of the technologies have started to converge in unique platform supporting a variety of services including processing, communication, entertainment and multimedia. There are many possible ways this technology convergence is developing, and many opportunities to integrate and deploy current technologies which could be exploited by hardware and software companies, Telcos and ISPs in partnership with the game industry.

One of the most striking convergences is that occurring within IPTV (Internet Protocol TV) industry [2, 3]. Many IPTV providers offer platforms supporting a variety of services including triple play services (e.g. TV broadcast or VOD, phone/VoIP and Broadband/Internet access). Some of them have also started to offer access to game portal allowing their users to entertain themselves with casual games (e.g. Othello, Sudoku, card games, Puzzles games, etc...) and even a few multiplayer online games. IPTV provides a new type of opportunity for the internet gaming. So far web-based multiplayer games have mainly been either casual single-player (or few players in the same) games in which results are compared, or massive multiplayer games and game environments mainly targeted at hard core gamers. Multiplayer gaming in IPTV however provides a new type of very casual gaming, in which all the players can play the same game and see this multi-gaming all the time. However there are still many games technologies which are not being fully exploited in IPTV.

In this paper we discuss the opportunity offered today by games technologies to facilitate game development, deployment and integration on IPTV platform. The challenge is to provide tools that simplify the production of 3D gaming and virtual environments to allow anyone to contribute to the production of the environment and the content within such environments. Another challenge is to support the convergence between existing and emerging technologies and enabling the deployment and integration of game

## IEEE COMSOC MMTC E-Letter

platform with other internet supported media such as IPTV. We have developed Homura, a platform designed to facilitate the development of 3D Java games [1, 4, 5] in an all inclusive interface, which provides text and graphical editors, compilers and launchers, and interfaces to objects in the game. We have utilized the advances made within many existing technologies to create a novel platform where all aspects of gaming can be achieved. This provides great benefits. First, it provides a single environment, which contains different development views dependent on what part of the game you are developing, thus the user does not have to learn and use many different applications. Secondly, it provides a set of simplified tools that can not only be used by professional game developers but also by less

experienced developers. In this way we can tap into the imaginations of anyone connected to the Internet.

The structure of the paper is as follows; in section 2, we discuss IPTV and key components; in section 3, we introduce Homura and NetHomura a game technology middleware suites for web-based Java 3D games development and deployment; in section 4 we propose an integration framework supporting deployment of NetHomura games in the internet and in an IPTV platform using Peer-to-Peer based architecture; and in section 5 we draw the conclusions and discuss future direction of the project.

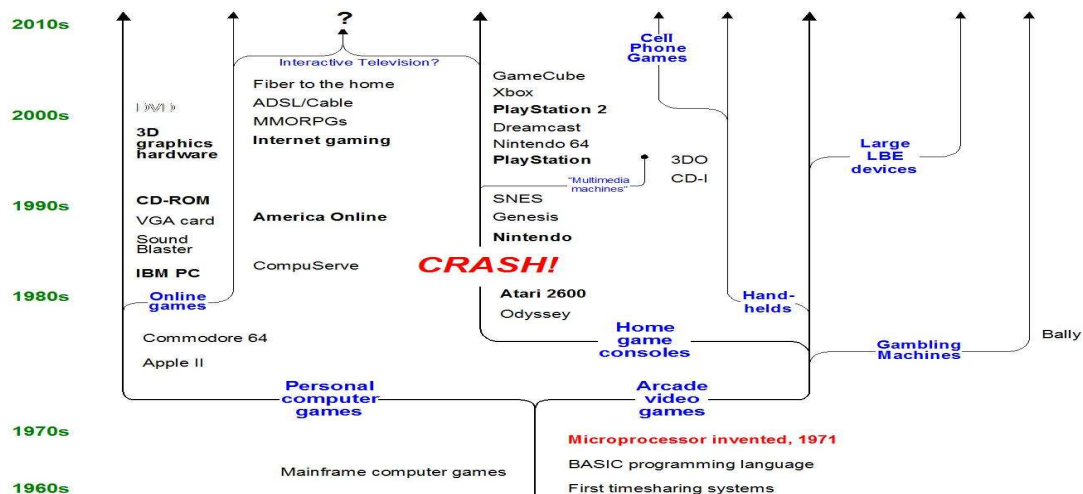


Figure 1: Game Industry Technology Development

## 2. IPTV Technologies

IPTV [3] is a technology for preparing and distributing Television signals over an IP based data network. Content streams are acquired, demodulated and decrypted if necessary, then re-encoded digitally for IP transport possibly, with additional compression and new encryption. IPTV signals, or streams, are then distributed on a network and viewed on an IPTV capable viewing device, usually a Set Top Box (STB). The key components of an IPTV system are Programming, Headend, IPTV Distribution Network and IPTV Viewing Device. [2]

### 2.1 OVERVIEW OF IPTV KEY COMPONENTS

#### 2.1.1 IPTV Programming

All programming or content is owned by some person or some entity. In this industry the entity that owns the rights to the content is called "The Programmer" and the IPTV Service Provider (IPTVSP) has to acquire the right to collect and redistribute this material.

#### 2.1.2 IPTV Headend

The Headend [2] is the physical facility where you collect the media streams and de-code/re-code as IP for distribution. This is the core of the IPTVSP's world. This facility is no different in

## IEEE COMSOC MMTC E-Letter

principal from a traditional Community Access TV (CATV) or Cable Television headend. The IPTV headend facility will differ from a traditional CATV headend in that once demodulated and decrypted IPTV will be Multicast encoded. That means the content stream will be digitally generated and ejected from its encoding equipment on a Multicast IP address.

The IPTV headend facility is typically also the home of a collection of important servers. These servers perform the various duties that make IPTV an enjoyable and usable experience to the end user and pleasant to the Programmer. Some but not all of these items are; Middleware server(s), Digital Rights Management (DRM) Server(s), Video On Demand (VOD) server(s), Storage Arrays, DHCP servers, STB image/boot server(s), Emergency Alert System(s), and any number of other servers or appliances which are necessary for the network operations of a high up-time, high bandwidth and robust network.

### 2.1.3 IPTV Distribution Network

The IPTV distribution network needs to have adequate bandwidth to the end users to deliver the content, and it needs to be capable of transporting IP Multicast based communications. An IP based data network that most people are familiar with is the Internet. Other examples of IP networks may be an office LAN, a home LAN or a Private WAN. These networks may connect to and give users access to the internet, but in practice are not "the Internet".

Building an IPTV Distribution Network can be attained with the mixing and matching of many current technologies. As emerging technologies become available to deliver more bandwidth to the home, these new technologies will certainly be incorporated and used as the IPTVSP's network engineers develop ways to use these new technologies most efficiently. [2, 3]

### 2.1.4 IPTV Viewing Device

The Viewing Device or Set Top Box is the adapter, which connects to the users' television. It is similar in appearance to the STB provided by the cable or satellite company for their digital services. The STB is the hardware item that the user associates with the IPTVSP, but it is useless without a middleware solution. IPTV signals can also be viewed on a PC; many software media players can play Multicast streaming content. But, to be a legitimate IPTV client a media player would have to interact with the Digital Rights Management package. There is nothing

the programmers fear more than for all their intellectual property dumped out onto the internet for a file-sharing abuse.

## 2.2 IPTV SERVICES AND INTERACTIVE ENHANCEMENTS

The delivery via the internet makes it possible to support a very large number of services beside television or video on demand. In fact a very common mode of services provided via IPTV is triple play where TV, phone and broadband are made available. Beside these services, new technologies made it possible to offer games on this platform.

One of the main attractions of IPTV is its ability to provide a much richer television experience through its addressability and interactive functionality. That interactive functionality includes:

- Provision of customized content both with streaming video and VOD. Instead of paying for channels that are rarely viewed, IPTV subscribers receive only those channels that they want.
- Communications capabilities such as video telephony, e-mail, chat, and other options are all available on the television screen
- Multiple Picture-in-Picture (PIP), with the option to chose any specific one and to change camera angles, follow a favorite sports player or watch a favorite car in a race.
- Home networking, where the viewer can control electronic devices within the home from the television screen or by remote web access.
- Real-time shopping from the television set.
- Networked Gaming, where viewer can play any game they want, when they want, with whomever they want.

## 3. Homura and NetHomura Platforms

### 3.1 Homura Libraries and IDE

#### 3.1.1 Homura IDE

Homura is a gaming integrated development environment based on the Eclipse Platform. This makes it extensible through the use of the Eclipse plug-in technology. One particularly important plug-in the IDE uses is JDT (Java development Tools), which provides the user with a rich Java editor for creating game logic. Various parts of Homura are declaratively specified in Extensible

## IEEE COMSOC MMTC E-Letter

Markup Language (XML) files located in the root of a project. This provides a link between classes and concepts (the main application class, for instance) and functionality exposed by and to other plug-ins. The IDE can act upon and manipulate these files to modify how the game functions. This is similar to how Eclipse works with the `plugin.xml` located in the root of each rich client program project.

The IDE itself hooks into the running Homura engine while a Homura application or game is running to provide various introspection and debugging facilities. For example, the user is able to see details about the concepts which are in operation within the application, as well as monitoring the current frame-rate through a statistics view and to inspect the scene graph view.

All the game examples presented in this paper have been developed using Homura, jME [6] and LWJGL [8], and integrated in Homura IDE. The game engine has been extended with libraries to support graphic and gaming functions. Using these set of libraries, users can create Homura projects and run them independently of the IDE. They can also be exported to a wide range of other platforms, making our solution highly open, cross-platform and extensible. An extended version of MonkeyWorld3D [9] has also been integrated, providing Homura with spatial and positional editor capabilities. We have drawn from many other contributions [10] to extend the capabilities of Homura, which include 3D engines and interfaces. This in its present form allows for the development of complete 3D Java games supporting advanced 3D game features, ODE based physics, and game assets such as Collada, 3DS or MD5 formats and textures [11][12]. The Homura engine is available in [12] for download as open source.

### 3.1.2 Homura Engine

The Homura Engine is based on an extended version of jME [6]. The extensions allow the internals of jME to be refactored, and optimised. This provides a closer integration with Homura data structures and classes. Using standard game functions in Homura we can parse over the various XML files created by the IDE to achieve different goals (e.g. loading models, managing scene graphs, or applying textures and applying special effects.)

### 3.1.3 3D Engine

Using the jME scene graph engine, the models which comprise a game can be rendered to the screen. The task of the 3D engine is to manage all 3D objects and scenes, and handle complex graphical interactions and effects. These

can either be handled in high-level or low-level. The higher level is assumed by the application itself with the help of the scene graph, and is managed at the software level. The lower level is assumed by the renderer and is managed at the hardware level with the help of 3D acceleration when available).

To put a two-dimensional raster on the screen from the information of three-dimensional objects, a virtual camera, textures and lights, and a rendering pipeline are used. The 3D engine is a wrapper with the given 3D interface, for example, OpenGL, which abstracts the rendering pipeline. The pipeline itself is most frequently implemented on 3D acceleration hardware and graphics cards, which are included in almost every modern computer. The pipeline is filled with 3D data which is changed while running through the pipeline. The result at the end of the pipeline is a complete picture on the screen. Common graphics features developed and used in Homura are comparable to those found in Java 3D and in the commercial Java engine Agency 9's AgentFX[1]. These include:

- Octree spatial sub-partitioning for sub-scene management, and polygon-based collision detection
- Key frame animation and skeleton based animation systems
- Dynamic Lighting, vertex shaders and pixel shaders
- Mip-mapping, bump-map support, along with basic and multi-texturing
- Support for major 3D modelling tools such as Maya, 3D Studio Max using Sony's COLLADA format.

### 3.1.4 Scene Graph

The main task of a 3D engine is to maintain and display 3D objects. A collection of 3D objects in a game is known as a scene. To relate all these objects, a so-called scene graph is used. In Homura, this scene graph corresponds to a tree with a root and other graph elements. Node elements can have multiple child elements but only one parent element while spatial elements cannot have child elements (Node extends Spatial to add this functionality.). The scene-graph aspect of Homura allows a large level scene graph to be created, without having to process it all at once. Only the regions that are being observed are of direct interest, while scenes that are further away have a reduced amount of observable detail in areas, such as meshes and textures. This general concept is commonly known as spatial subdivision, with the graphical aspect being known as level-of-detail (LOD) management. The trade-off for this reduction in processing is

## IEEE COMSOC MMTC E-Letter

more memory usage, and more code required for these aspects of the engine.

### 3.2 Supporting Technologies

#### 3.2.1 *Integrated use of Java and IDE*

We have chosen Java as the primary language within our project. By using the same implementation language and runtime for all the game development systems (i.e. engine, IDE and games) we improve the amount of compatibility and interoperability between the various sub-components we are developing. For instance, by having the engine and IDE both developed in Java, we allow the engine to run inside the IDE and allow the user to introspect upon various parts of the engine, and debug the games during execution. This also aids multi-platform execution and reduces the overall complexity of game development in Java.

#### 3.2.2 *The Eclipse Platform*

The Eclipse Platform is being used as the basis for our Homura (IDE). It has proven itself to be a very mature platform for tools development in many different areas of computing. Using this platform, we can integrate various existing Eclipse plug-ins and use them in different ways. For example, the Java editor plug-in provides the user with a rich Java development environment, which is highly suitable for advanced games programming, providing that we have a rich Java API for game development, like that provided by jME. In the project the target development platform for 3D games development is web-based. However, through IDE extensions, games can run as a standalone application, an Applet or a Webstart application capable of running on different platforms such as Windows, Linux, Linux on PS3 (with YDL or Fedora Core 5) and deployed as web-based games or on IPTV.

### 3.3 The IDE and General Tooling

#### 3.3.1 *Project Setup*

The project setup wizard in the Homura IDE allows the user to set up a Homura project quickly, by including all the necessary Java libraries and setting up the file system hierarchy. During setup, the initial XML files are created, which describe the static aspects of the game, and

the linking elements between the engine and the IDE. These XML files will become more important in the future when we want to link Homura to a wider internet community, which would allow users to export their game creations to a website using the various export wizards provided by the Homura IDE. This would also facilitate user-generated content. The server-side software will be able to extract these XML files from the exported Java archive, and then obtain the various details of the game, which can be described on the game page.

#### 3.3.2 *The Java Editor*

The Java editor is an important element of the IDE, allowing the user to actually develop the core game logic in Java. This is important for the engine to be highly dynamic, and not restrict the user in what they can do. We are looking into supplementing the editor with new features that aim to accelerate Java game development, such as the creation of standard templates for pieces of boilerplate code that are used frequently when creating jME and LWJGL games. For instance we provide Homura-specific templates such as loading spatial and sound. Using the cheat sheets feature and template, it is also possible to create step-by-step examples for creating games using Homura.

#### 3.3.3 *The Spatial Editor*

The spatial editor, shown in Figure 2, allows the user to have discrete control over how the node hierarchies of individual objects in the world are constructed, allowing the user to manipulate the camera, objects, materials, textures, and lighting of each individual spatial hierarchy. To get a spatial editor into our IDE, we use classes extended from MonkeyWorld3D. The idea is to integrate this editor, ensuring a synergy between the spatial editor development and the other components of the Homura IDE, such as the Java editor and assets browser, and include the required rendering classes.

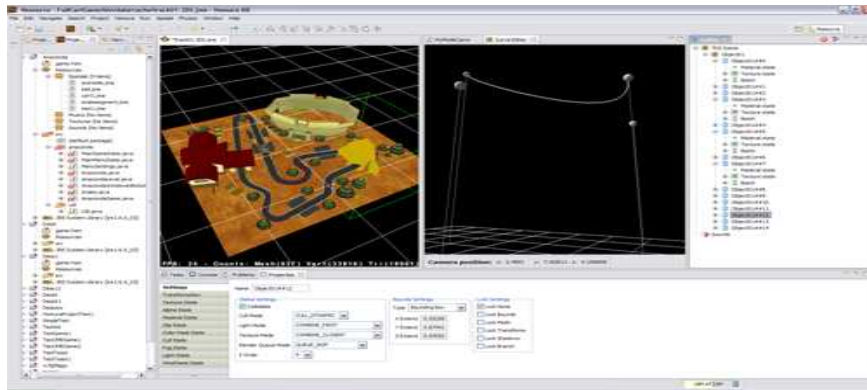


Figure 2: Homura IDE Interface



Figure 3: “The Islands of Dr Bubbles” Game Screenshot



Figure 4: “Cart Game” Screenshot

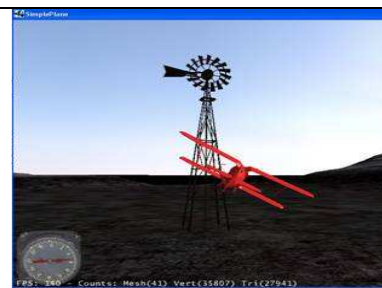


Figure 5: “Biplane Game” Screenshot

### 3.3.4 The Positional and Curve Editor

The positional editor is based code from the spatial editor, and allows the user to place objects in the world using an object transformation storage system. The underlying implementation uses a hash map system where the user puts an object ID in, and gets positional and transformational information out about the object in question. It allows the user to have some control over how they use the positional information, rather than some automated system which uses it for them. This is useful for map editing and game logic integration. Alongside the aforementioned functionality, the curve editor allows the user to import existing curves and create curves from scratch which can be exported. We found that this functionality was required in the production of the games from developer’s feedback.

### 3.4 Java Game Design and Development

We have developed several games Homura components. The games shown in Figures 3-5 are examples of such development. These games utilise advanced features such as height map based terrain pages with procedural multi-texturing and detail/bump mapping [7], multiple light sources, and scene graph-based modal representations with frustum and back-face culling. For the collisions, we use an oriented

bounding box / sphere collision detection system, with the ability to upgrade to a ray traced triangle pick method for more precise intersections. Other features include shadow render passes, dynamic sky domes, camera animation, and 3D model loading in several different formats.

## 4. NetHomura Deployment and Framework for IPTV

### 4.1 Networking Middleware and Online-Deployment Mechanisms for Java-based Games

Online Games portals (such as EA’s Pogo [13] and Habbo Hotel [14]) principally use the proprietary Adobe Flash technology to distribute web-based and downloadable game titles through web browsers or IPTV. Other proprietary technologies are the *Unity* game engine [15] and *Instant Action* [16]. They provide browser based play for Windows and Mac OS X via its custom plug-in for Internet Explorer, Firefox and Safari. In comparison to Flash, they provide a much more powerful framework for online games, enabling hardware-accelerated 3D games using DirectX and Open GL to embed within a web page and execute directly within the browser window, or in full screen mode.

The Homura games project [12] investigated the development of a cross-platform games engine

## IEEE COMSOC MMTC E-Letter

and Integrated Development Environment (IDE) for the development of hardware-accelerated 3D Java games using open-source technologies, based on jME [17][18]. The following sections aim to extend the functionality provided by the Homura Engine, adding systems catering for both the deployment and networking of games built upon the platform, via existing web technologies. The deployment platform provides a game portal web application, which facilitates authenticated access to the Homura-based games and securely deploys the games to the client computers via the Internet. The portal offers features such as automatic updates and the ability to easily enhance existing deployment through additional game content. In addition to the distribution mechanism, the portal also provides an interface which allows users to search for other games, create friendships with other members and instantaneously receive communications regarding the release of new titles on the portal. The system manages game data from in-game experiences such as high-scores and game specific statistics. The networking middleware platform allow developers to easily create Homura-based games which can offer multiplayer, online gameplay in a cost-effective manner leveraging the resource sharing capabilities of the P2P networking topology, whilst still retaining centralized control over access rights and the maintaining the integrity of persistent game data. This section of the paper describes the motivations for the project *NetHomura*, the initial design and analysis phase and the architectural structure of the resulting prototype, named *NetHomura*.

### 4.2 Java Deployment

Currently, there are two methods of deploying Java-based applications via the internet, Applets and Java Web Start. The Java Web Start technology [19] was chosen as the base platform for the deployment system as it offers several key advantages over using Applets. Java Web Start (JWS) is a standard component of the Java Standard Edition Development Kit (Java SE JDK) and is distributed as a part of the Runtime Environment (JRE). The JRE is required by users in order to execute Java Applications and provides a browser plug-in. The Java Plugin is freely available for all major operating systems and browser environments, making the technology ubiquitous amongst desktop PC users. JWS utilizes the Java Network Launching Protocol (JNLP) [20] to configure exactly how an application is deployed from a server location to the clients' machines. The protocol uses standardized XML schema to define several key aspects of the deployment process.

- The appearance of the download window presented to the client upon deployment initiation.
- The creation of desktop and system shortcuts.
- The off-line availability of the application.
- References to any external Java Archive (JAR) files required
- Support for Operating System and architecture specific (e.g. x86/x64/PPC) libraries.
- Dynamic Properties passed as arguments to the application.
- The permissions and security access provisioning of the application.
- Specification of the entry point of the application.
- The configuration of resource updates from the initial deployment location.

#### 4.2.1 Web Start Features and Advantages

An Applet is a specialized Java application, which is embedded directly into the HTML of a web page, and transferred and executed directly on the client machine, within the context of the browser. The browser integration of Applets inherently limits the ability to run full-screen applications, and also restricts the resolutions at which games can operate. With Applets, the Java Virtual Machine is invoked directly as a process within the browser, and multiple applets must share a JVM instance, which carries a performance and resource overhead.

JWS applications are executed completely independent of the browser, and each instance is executed within a separate Java process with performance equal to that of standard Java applications. The browser independence means that the application can be adjusted to a variety of resolutions and can run in windowed or full-screen mode with no overheads. An application can be configured by the developer or system administrator to consume additional resources by adjusting the JVM's start-up configuration through the JNLP specification. JWS applications can be easily integrated into HTML pages via a link to the JNLP file, which when executed, invokes the cross-platform Java plug-in, subsequently handling the execution of the application. As a result, JWS exhibits excellent cross platform, cross-browser support. In comparison, the inconsistencies found with the implementation of Applets across browsers and Operating Systems makes the creation of a consistent user experience a difficult task.

## IEEE COMSOC MMTC E-Letter

With JWS, the installation of native libraries onto the end user's machine is seamless and eliminates the complexities which can be encountered when attempting the same process using Applets. This is a crucial factor, as LWJGL and consequently jME and Homura rely on native libraries for supporting OpenGL, non-standard input devices such as gamepads and joysticks and also for hardware accelerated audio capabilities provided by libraries such as OpenAL and FMOD.

### 4.2.2 Networking for Java-Based Games

Java features an extensive networking API incorporated into the JDK, which many additional networking solutions APIs are built upon, including those designed for Java games. Two such examples of these are Sun's *Project Darkstar* [21] and the jME-affiliated *Java Game Networking* (JGN) [22] which are both open source. Both of these libraries provide game-specific Client-Server APIs, and can be integrated with jME, and therefore Homura.

Client-Server architectures are popular in game networking, because they are relatively simple to implement, allow the developer to maintain absolute control over the state of game and the use of a centralized authority makes cheat-detection and user access control a simple task. However, scalability is an issue, as it is often inflexible and also extremely costly due to the amount of dedicated hardware required to sustain such design. The network has to be over-provisioned to handle peak loads and limits the deployment of user-generated content [23]. Client-Server games may also have limited life-spans, as they will only remain active whilst the server infrastructure is still supported. The issue of hardware costs means that using Client-Server architecture to support many games is infeasible for small-scale developers, and would create a barrier of entry for using this platform. A Peer-to-Peer (P2P) solution would allow the system to utilize the resources of the connected users (CPU, Memory and Network Bandwidth), creating an easily scalable system and one that is in keeping with the aforementioned ethos. Therefore, the Java version of the JXTA [24] P2P platform (JXSE 2.5) was selected as the underlying networking library for the middleware solution. JXTA is an open source initiative created by Sun Microsystems in 2001 to standardize and provide a set of P2P protocols. JXTA comprises of a set of six protocols that support common P2P networking tasks, such as peer discovery, organization of peers, peer identification and verification and messaging. The JXTA protocols [25] use the open standards of XML schemas to define their semantics and subsequently allow

JXTA to operate independently of language, operating system, network topology and even the underlying transport protocol. The Java JXSE version is the reference implementation of the JXTA standards. JXTA does not preclude the usage of Client-Server topologies, and persistent peers can be utilized to great effect. It neither excludes nor inherently depends on centralized control points. JXTA has been used as the basis for many recent P2P studies, and its feasibility for use within game-related network programming (even large scale applications, e.g. MMOGs) is proven [26][27].

### 4.3 System Overview

The *NetHomura* deployment platform and networking middleware is still evolving. This paper presents the work completed during the preliminary phase of the project. This first stage includes the implementation of the deployment platform, and the initial API for the creation of multiplayer online Homura-Based games. In this section: we present the combined architecture of the two technologies and the interactions which exist between them; we explain the deployment process, how it can be used to rapidly and securely deliver feature and content rich games via standard web technologies; and we discuss the infrastructure of the network overlay which the client applications will form during the course of a networked game session.

#### 4.3.1 Deployment Platform Overview

The deployment solution provides a web application which provides a portal for access to games built using the Homura platform. The deployment portal provides the following features to the user and developer:

- User Accounts/Authenticated Access
- Game Catalogue Facilities
- Game Statistics Storage
- Caching and Update Facilities
- Logging and Access Control

#### 4.3.2 Networking Middleware Overview

The networking middleware provides a high-level implementation of the JXTA networking facilities, allowing the game developer to incorporate networking easily into their application. The middleware currently provides the following features:

- 8-16 Concurrent Players within a single game instance.
- Creation of a network connection from a peer.
- Creation and joining of Peer Groups.
- Sending and Receiving Game-Related Messages.



## IEEE COMSOC MMTC E-Letter

- Communication of Shared Data to the data repository.
- Representations of users and game session as Java Objects.
- Monitoring of peer latency and bandwidth across the session.

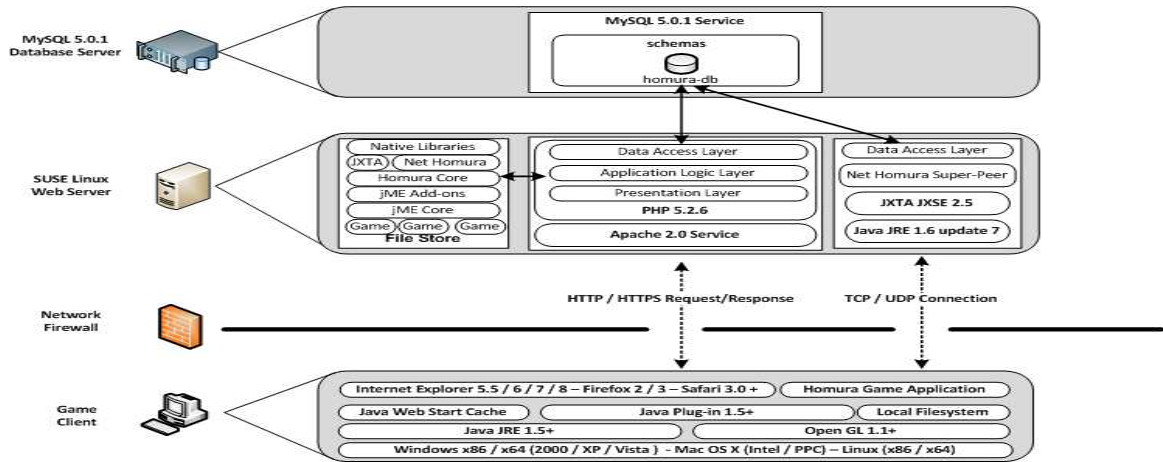


Figure 6: NetHomura System Architecture

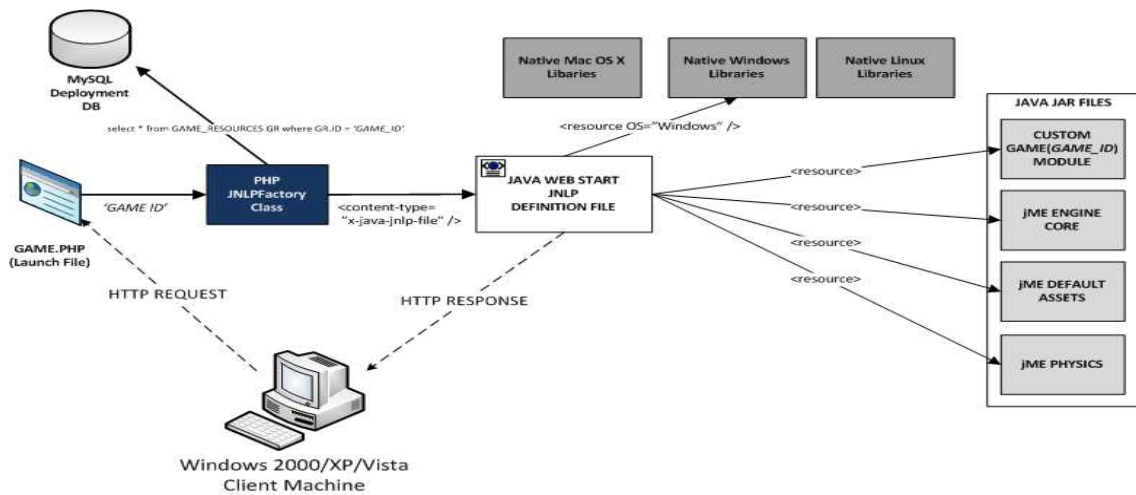


Figure 7: Deployment Process using Windows

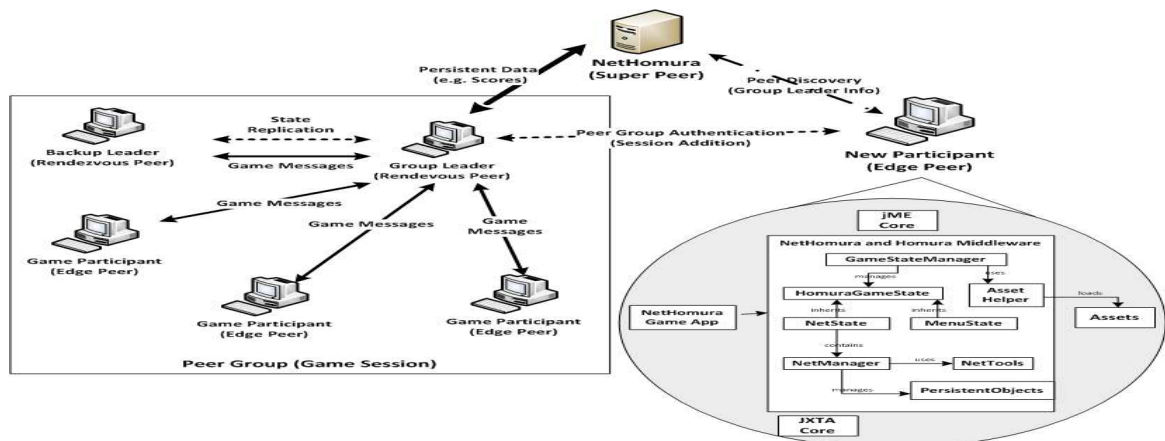


Figure 8 : Sample P2P Game Interactions via NetHomura

#### 4.4 System Architecture

NetHomura utilizes two physical servers to host the solution: a SUSE Linux, Apache 2.0 combined web and application server running PHP 5.2.6 and a Database Server running MySQL 5.0.1. This forms a LAMP (Linux-Apache-MySQL-PHP) configuration, which is commonly used for standard web applications. The application is extremely portable and would work equally well on a Windows-based machine using either Apache 2.0 or IIS. The use of two servers is due to the internal infrastructure available and could easily be combined into a single server. Figure 6 provides an overview of the architecture, illustrating the physical and logical layout of the application and the communications permitted between the server and the game clients.

The deployment solution is programmed using a blend of Object-Oriented and Page-Oriented PHP, and is separated into a three-tiered architecture:

**Data Access Layer:** The solution uses MySQL for its persistent storage. A PHP Data Access layer has been implemented, which abstracts the database from the application, allowing the system to retrieve application class instances representing the database objects, and handles the insertion or modification of data within the database based on the current state of the application class. This abstraction also means the database can be replaced with another RDBMS (e.g. Postgres), or use another solution, such as XML files to store the data.

**Application Layer:** The Application layer provides classes which represent the data in the database and encapsulates the application logic. The system current features implementations of

users, games, online game modes; game scores etc. The class instance data is modified by actions carried out by the user via the application interface, using a simple getter/setter pattern.

**Presentation Layer:** The solution programmatically constructs the HTML and JavaScript-based page template and displays dynamic views of the system to the user, using the classes of the Application Layer. This abstraction means that the entire look and feel of the application can be replaced without affecting the application logic and allows the system to incorporate AJAX/DHTML libraries to enhance the user experience.

The games and supporting libraries (Homura, jME, Native DLLs/SOs) are stored on the server's local file system and are not directly accessible from outside of the server. The files are stored as JAR files and are compressed using the g-zip compression system. This minimizes the size of the game distribution. The partitioning of the JARs means that, once a client has downloaded their first game the Homura framework is installed, and that any future game installation will only require the game JAR to be distributed, further reducing installation times. The JARs are signed using digital signatures so their integrity and authenticity can be verified, combating binary modification (commonly used for online cheats). This also means that the application does not have to run within the standard Java sandbox, providing additional permissions to the application, such as running Native Libraries (e.g. OpenGL), storing data locally on the user's machine, and creating outbound network connections (used by the middleware). The Server also hosts a persistent JXTA Super Peer, which can be accessed by any

## IEEE COMSOC MMTC E-Letter

of the Clients running NetHomura based games. This Peer is used to advertise the location of other games users, to insert/modify any persistent data (scores, logging information, user-identifiers). This peer is used to allocate network functionality to connected peer applications, delegating more and more power to the peers, and acts as a proxy, so that peers cannot directly access the database. Both the deployment and middleware provide multi-platform support. A game client can currently use Windows, Linux, or Mac OS X Operating Systems and Internet Explorer, Firefox or Safari browsers and requires OpenGL to be installed on the system.

### 4.5 Deployment Process

Java Web Start applications are usually deployed using a static JNLP file with a reference link embedded within an HTML page. However, this method is unsecure, and means that anyone with knowledge of the location of the JNLP file can access and deploy the game. Thus, the deployment architecture dynamically creates the JNLP file to protect the games from unrestricted access. Figure 7 illustrates this process for a Windows machine running either Firefox or Internet Explorer.

A user who is authenticated into the deployment portal can locate a game via the search mechanism, and view the games details (download size, multiplayer modes, online support, screenshots etc.). The application provides a 'Launch' button, which then invokes the deployment of the game. The game's unique identifier is passed to the JNLPFactory class of the deployment application layer. This class queries the database via the data access layer to retrieve the required resources of the game, JVM settings, properties (used to pass user-specific data to the applications) and dynamically constructs the JNLP file, which is returned to the client via an HTTP response. The browser then passes the JNLP file to the Client's Java Plugin which parses the JNLP file and downloads the game application and resources associated with the application. The Client's architecture and Operating System is established by the Plugin so that only the Native Libraries relevant to the client are downloaded. Once the game is cached, it is then executed using the Client's JVM. The process is similar for subsequent executions. When the JNLP is sent, the contents of the JWS cache on the client's machine is analyzed and differentiated against the server copies. If new versions of any of the associated JARs, or any new JARs added to the application (e.g. new level packs, assets) then these are downloaded and returned to the client.

### 4.6 Peer-to-Peer Networking

The networking middleware of NetHomura utilizes the key concepts of the JXTA framework, and builds upon them for the purposes of gaming, with the resulting network created by the framework is a hybrid P2P topology, as illustrated by Figure 8, which details both the network structure and the software architecture of the peers within the system.

#### 4.6.1 Middleware Overview

The NetHomura middleware integrates with the Homura Engine to create a *GameStateManager* to control the game. This manages an internal stack of *HomuraGameState* instances. *HomuraGameState* is an abstract class which implements the game loop of each state, providing methods for initialization of content, handling user input and updating the state of the game world (members of the scenegraph), and a rendering the scenegraph to screen. The NetHomura games are comprised of concrete implementations of this class (e.g. *MainMenuState*, *LoadingState*, *PuzzleGameState*, etc.). The middleware provides an additional implementation, *NetState*, which encompasses the additional interactions of a network game, by adding methods to receive messages, send messages, join and leave games. The *NetState* class uses an instance of the middleware's *NetManager* class, which handles peer-management facilities such as discovering available game sessions, creation of new game session, tracking and modifying persistent, shared data objects used within the game, managing references to connected peers, sending messages to particular peers and retrieving messages that are received from peers.

The *NetManager* also handles session control, such as disconnecting and joining into both the entire network and game sessions. The role of the game developer using the middleware is to create game-specific messages which inherit from the base *NetHomuraMessage* class. This class encapsulates the in-game messages sent between peers, and using the *NetTools* class to construct efficient managements using the functions to efficiently serialize Java object into messages. These messages can then be broadcast using the *NetManager*. The middleware also provides the concept of *GameSessionAdvertisements*, which are used to create and communicate the details of a particular game session to other peers so that they can participate in a session.

#### 4.6.2 Network Topology

Games built using the NetHomura framework construct a hybrid P2P topology. Initially, the NetHomura *Super Peer* on the deployment server

## IEEE COMSOC MMTC E-Letter

forms a Client-Server network, which acts as the gateway to the P2P game sessions, authenticating the user based on their deployment credentials and transmitting the location of any sessions available for the user's game. A game session is organized as a *Peer group*. The Peer group is assigned a group leader, which becomes a *JXTA Rendezvous Peer* (initially the creator of the group). The Rendezvous peer is responsible for broadcasting the services offered by the group (e.g. the game session) using JXTA's advertisement framework. The rendezvous peer is also responsible for propagating messages received from other peers within the peer group to all other members [25], who are connected as *Edge Peers*. Edge Peers have a single responsibility, to communicate their current game state to the group leader. A replica of this peer is also created to provide redundancy. If the group leader disconnects, then leadership is switched. Peers are connected using *JXTA Socket* connections, communicating via TCP/UDP. A secure connection using TLS is created between the group leader and the Super Peer to securely transmit persistent data, for entry into the deployment database.

### 4.7 Deployment Framework for IPTV

Homura will offer gaming-on-demand for broadband networks. In the next stage of the Homura project a dedicated cluster of game servers will enable the 3D games to be played on demand on any broadband-connected device, including existing cable and IPTV set-top boxes and 3G mobile phones. In the case of cable and IPTV deployments, Homura gaming-on-demand will allow network operators to leverage investment already made in video-on-demand (VOD) infrastructure. All existing and future 3D java game titles could be brought to the Homura platform with minimal porting work. Also the peer-to-peer -centric nature of the system will be particularly well-suited to multiplayer and other community-building games.

### 5. Conclusion

Currently, web-based online gaming applications on IPTV are predominately utilising Adobe Flash or Java as their core technologies. These games are often casual, two-dimensional games and do not utilize the specialist graphics hardware which has proliferated across modern PCs and Consoles. Multi-user online game play in these titles is often either non-existent or extremely limited on IPTV platform. Computer games applications which grace the current generation of consoles and personal computers are designed to utilize the increasingly

impressive hardware power at their disposal. However, these are commonly distributed using a physical medium or deployed through custom, proprietary networking mechanisms and rely upon platform-specific networking APIs to facilitate multi-user online game play. In order to unify the concepts of these disparate styles of gaming, in this paper we proposed a novel integrated development environment called Homura and NetHomura, and proposed and framework for its deployment. Homura is based on the Eclipse platform and extends the jME game engine, with new interfaces, content and libraries, thus, providing a software suite that integrates source editors, compilers, including spatial and positional editors to afford advanced graphical functionalities within the IDE. We also presented two interconnected systems which are implemented using Java Web Start and JXTA P2P technologies, providing a platform-independent framework capable of deploying hardware accelerated cross-platform, cross-browser online-enabled Java 3D games, as part of the NetHomura Project. Future work involves the deployment of the Homura platform on IPTV.

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## IEEE COMSOC MMTC E-Letter

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Editor's Selected Book Recommendation

Jenq-Neng Hwang, *Multimedia Networking: From Theory to Practice*, Cambridge University Press, 544 pages, April 2009.



**Overview**

Numerous multimedia networking applications have matured in the past few years, ranging from distance learning, to desktop videoconferencing, instant messaging, workgroup collaboration, multimedia kiosks and entertainment. Generally, there are four major components in multimedia networking. The first one is multimedia data (e.g., speech, audio, image, and video) compression that supports different end terminals and sometimes supports interoperability. The second one is the quality of service (QoS) issues, which include packet delay, packet loss, jitter, etc. The third one is the communication protocols for heterogeneous networks, which confirms the ability to keep the QoS in network convergence. The fourth one is to ensure the multimedia-networked content fully interoperable, including the content management, interoperation for delivery, and intellectual property management and protection, etc.

This book, authored the experienced expert, focuses on the four major components, and presents both theoretical analyses and practical applications. It provides a complete system design perspective based on existing international standards and state-of-the-art technologies, and presents an ideal balance of

theory and practical design and integration knowledge. Specially, it shows the designed multimedia coding and networking laboratory contents that may provide students with practical and hands-on experience in developing multimedia networking systems. The coverage and materials of this book are very appropriate for graduate students and researchers.

**Book Content**

The book is composed of 12 chapters. Chapter 1 gives an introduction to multimedia networking, including the history of multimedia networks, present multimedia communication systems and services, and major components of multimedia networking. Chapters 2-5 cover the first major component of multimedia networking, i.e., multimedia data compression. Here, the standardized encoding/decoding methods with respect to four types of multimedia content are considered, i.e., speech, audio, image and video. These popular compression standards are introduced and their performances are tested and compared. Chapter 6 introduces and discusses several popular digital multimedia broadcasting systems widely used in the world, e.g., Digital broadcasting services (e.g., Digital Cable for

## IEEE COMSOC MMTC E-Letter

Enhanced Definition TV (EDTV) and High Definition TV (HDTV) broadcasting, DirectTV via Direct Broadcast Satellite (DBS) services, and Digital Video Broadcasting (DVB), etc.). Chapters 7 and 8 focus on the Quality-of-Service (QoS) techniques for multimedia streaming over IP networks, ranging from MAC, network, transport and application layers of IP protocols. In detail, several commercially available multimedia streaming systems are also investigated. Chapters 9 and 10 discuss the advances of wireless broadband technologies and the QoS challenges of multimedia over wireless broadband infrastructures and from several layers of IP protocols. Chapter 11 focuses on the Digital Rights Management (DRM) technologies for multimedia networking and the related standardization efforts. In Chapter 12, many development software samples for multimedia data capturing, compression, streaming for PCs and mobile devices, as well as GUI designs for multimedia applications are presented to provide readers better learning experience of multimedia networking.

### Conclusion

This book covers various important components of multimedia networking, and is well organized and clearly presented. It is a high-quality reference on multimedia networking. I would recommend this book to undergraduate and graduate students and researchers in electrical engineering and computer science, and also to practitioners in the communications and networking industry.

*\* The Column Editor recommending this book is Dr. Shiguo Lian.*



**Shiguo Lian** got his Ph.D. from Nanjing University of Science and Technology in 2005. He was a research assistant in City University of Hong Kong in 2004. He has been a research scientist with France Telecom R&D (Orange Labs) Beijing since July 2005. He is the author or co-author of more than 60 refereed international journal and conference papers, and book chapters. He has held 16 patents, authored and co-edited 5 books, e.g., "Multimedia Content Encryption: Techniques and Applications" (CRC Press, 2008), "Handbook of Research on Secure Multimedia Distribution" (IGI Global, 2009), "Intelligent Multimedia Transmission: Techniques and Applications" (Springer, 2009), and "Intelligent Multimedia Analysis for Security Applications" (Springer, 2009), etc. He got the Nomination Prize of "Innovation Prize in France Telecom" in 2006, and "Top 100 Doctorate Dissertation in Jiangsu Province" in 2005. He has co-edited several special issues for international journals. He is the organization member of some conferences, e.g., MINES2009, MobiSec2009 and MUSIC'08. He is the TPC member of refereed conferences, e.g., IEEE ICC2008/2009, IEEE GLOBECOM2008-2010, IEEE CCNC2009, IWDW2008, etc., and invited reviewers of refereed magazines and journals, e.g., IEEE Communications Magazine, IEEE Transactions on Image Processing, Computer Communications, etc. His research interests include multimedia communication and security, intelligent computing, ubiquitous communication and services.



Editor's Selected Paper Recommendation

T. Liu, Y. Wang, J. M. Boyce, H. Yang, and Z. Wu, "A Novel Video Quality Metric for Low Bit-Rate Video Considering Both Coding and Packet-Loss Artifacts," *IEEE Journal of Selected Topics in Signal Processing*, Vol.3, Issue.2, pp 280-293, Apr. 2009.

Recent advances in both wired and wireless networking technology have prompted the explosion of various networked video services, available to users through a diverse set of access links. To properly design and provide a video service, it is of paramount importance to accurately assess the video quality perceived by the end users. Subjective evaluation may be the only way to attain the quality assessment closest to the "true" value; however, it is tremendously costly to execute, and may not be practical in some circumstances. As a result, an effective objective and automatic video quality metric is of significant interest.

In a networked video application involving low bandwidth and unreliable links, received videos can be greatly degraded due to both lossy compression [1] and packet loss [2]. Therefore, when designing an perceptual video quality metric, many factors need to be taken into account, including (a) video encoder settings (e.g., compression algorithm, quantization parameter and the groups of pictures (GOP) length), (b) packet loss attributes (e.g., number, duration, and location of loss-affected segments): owing to the use of spatial and temporal prediction adopted in the compression technology, and (c) video content. The interactions among these factors render the development of accurate quality metrics a highly challenging problem.

One of the main contribution of this paper is to *jointly* consider the coding artifacts and packet losses simultaneously. The authors in this paper focused on low resolution and low bit-rate videos coded following the H.264 standard without bidirectional prediction. In their work, the authors first investigated the impact of individual loss attribute on the perceptual video quality, including the loss severity (measured by the error in recovering the loss-affected frames), the length of loss-affected segment (i.e., the error propagation duration after a loss), the loss location within a testing clip, the number of losses, and the loss pattern (spread vs. clustered). They found that the joint effect of loss severity,

error length and loss number can be captured well by the sum of PSNR drops in all loss-affected frames with some non-linear clipping, where the PSNR drop of a frame is the difference between the PSNR of the encoded frame and that of the reconstructed frame. They also observed that the impact of the loss location can be characterized by a forgiveness factor that decays exponentially with the distance of the last erroneous frame to the end of the clip. Finally the loss pattern effect was captured by a function of the inter-distances between losses, the loss span and the loss number. Based on these findings, they proposed an objective metric for the quality degradation due to packet losses that considers all these factors.

The authors also adopted and validated a prior metric that predicts the quality degradation due to compression artifacts using a sigmoidal function of the PSNR. The final proposed metric for the overall quality degradation due to both compression and packet loss is a weighted sum of these two metrics. The overall metric correlates very well with subjective ratings, for a large set of video clips with different loss patterns, coding artifacts, and scene contents.

Further study on video quality assessment on scalable video codec with temporal and quality scalability can be performed to help rate/quality adaptation in streaming application [3].

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\* *The Column Editor recommending this paper is Dr. Guan-Ming Su.*

## Enabling Broadcast of User-Generated Live Video without Servers

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### I. User-Generated Live Video

User-Generated-Content (UGC) has become tremendously popular on the Internet in recent years. The global connectivity provided by the Internet makes it extremely easy for users to share a wide variety of UGC, including blogs, photos and video clips. YouTube [1], the popular UGC video streaming site, serves 100 million distinct videos and attracts 65; 000 uploads daily. While Youtube offers pre-recorded video, in the upcoming years, we expect to see the emergence of *User-Generated Live Video*, for which any user can create its own temporary live video channel from a webcam or a hand-held wireless device. The live channel could be a professor's lecture, a little-league baseball game, a wedding, an artistic performance, or a political demonstration. Unlike prerecorded video, live video streaming has to meet the stringent video playback deadlines. The dissemination of user-generated live video to a large number of audiences is a challenging problem, and is the focus of this work.

### II. Possible Approaches with Servers

One may naturally resort to a server-based solution and build a live version of YouTube. Server farms and CDN networks can be employed to host UG live channels and stream videos directly to viewers. However, the server and network infrastructure cost grows proportionally to the number of viewers. YouTube is estimated to pay over 1 million dollars per month to content distribution network (CDN) for distributing videos [2]. Similarly, it will be very expensive to host a large number of UG live channels. Peer-to-Peer (P2P) technology has recently been adopted to offload servers by efficiently utilizing the upload bandwidth of end users. Dozens of commercial P2P video streaming systems, such as PPLive [3], have been deployed on the Internet to deliver live and on-demand video services. Although existing P2P solutions can offload server efficiently, a reasonably large server bandwidth is still assumed to achieve good streaming performance, such as low startup delay and chunk loss ratio, in large scale P2P streaming. Current commercial P2P IPTV systems still invest considerably on

servers that track peers, host video and bootstrap P2P video sharing. Technically, it is possible for those companies to provide a service to host UG live channels in the near future. However users and companies will have to reach agreement on various economical, legal, copyright and content control issues to make it happen.

### III. Pure Source-based Solution

*In this work, we investigate a "pure" P2P streaming solution that enables a user to broadcast his/her UG live video to a large number of audiences without any server support.* In our solution, any user generating a live video can act as a video source and directly drive a P2P live broadcast to users interested in his/her video. Unlike a commercial video streaming server, the user generating the video only has an upload bandwidth slightly higher than the encoded video rate and can barely send out one complete video stream to the rest of the world. The challenge is how to drive a small, medium or even large scale P2P live streaming using such a constrained source. We propose two strategies to address this challenge. First we investigate the impact of source-side chunk scheduling on the system streaming performance. We show that, by simply switching the source-side chunk scheduling from *passive* to *proactive* mode, a source with upload bandwidth slightly higher than the streaming rate can drive a small or medium scale P2P live broadcast. In P2P streaming, a certain amount of video playback delay on peers is necessary to facilitate video sharing. The playback delays tolerated by peers in current commercial P2P IPTV systems are on the order of tens of seconds [4]. Leveraging on peer's playback tolerance, we propose a Layered P2P Streaming (LPS) architecture that introduces peer playback delay differentiations and constructs virtual servers out of peers to drive large-scale streaming. In LPS, small fractions (e.g., 5%) of peers are assigned to an *amplifier* layer, and the rest of peers are assigned to a *base* layer. Peers at the amplifier layer have shorter target playback delays than peers at the base layer. The source only serves a small number of amplifier layer peers, which will then forward downloaded video to peers at the base layer. Effectively, there are multiple

## IEEE COMSOC MMTC E-Letter

virtual “servers” from the amplifier layer that collaboratively deliver a high level of video Quality-of-Experience to a large number of peers at the base layer. Fig. 1 shows a simple example of the architecture. A bandwidth constrained source distributes nearly one copy of the video stream to the peers at amplifier layer. Peers at the amplifier layer have abundant bandwidth and distribute one copy of data received to peers at the base layer, which then broadcast chunks to other peers at the base layer. In this case, the peers at the base layer receive three copies of data at streaming rate  $3r$ , and the source bandwidth has been amplified three times.

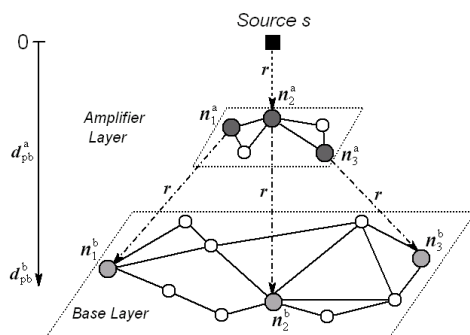


Figure 1: LPS Architecture

To support UG live channels, our work focuses on the scheduling and system architecture design to enable source with constrained bandwidth to stream video to a large number of peers. The proposed LPS architecture is the first one to deliberately differentiate the playback delays of peers in the same channel. Through extensive trace-driven simulations and experiments, we demonstrated that the proposed strategies enable a “weak” source with upload bandwidth slightly higher than the encoded video rate to drive a P2P streaming session with tens of thousands of peers. By sacrificing a little bit the playback delays for peers at the base layer, the proposed LPS architecture can achieve short start-up delays and low video chunk loss ratios for all peers at both the amplifier layer and the base layer.

Due to the space limit, we refer interested readers to our technical report [5] for more details.

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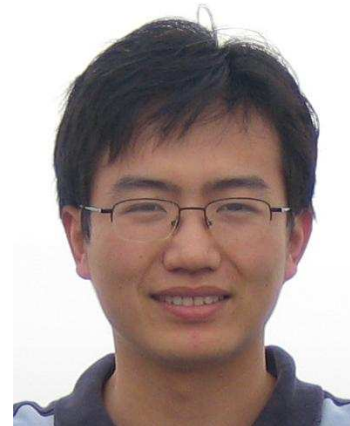
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## IEEE COMSOC MMTC E-Letter



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**IG Corner: Seamless Quality of Service Interworking across different Access Technologies**

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The interworking between different access technologies and, in particular, between 3G cellular and Wireless Local Area Networks (WLANs) has been considered as a suitable and viable evolution path towards the next generation of wireless networks. Yet, this interworking raises considerable challenges, especially when we demand for seamless continuity of multimedia sessions across the two access networks. These interworking challenges, and especially the QoS interworking, have been the main focus of the QoS Interest Group (QoSIG) for the last couple of years.

Typically, the 3G/WLAN interworking requirements are specified and categorized in terms of several usage scenarios. For example, a common usage scenario is when a 3G subscriber is admitted to a WLAN environment by re-using his regular 3G credentials, and then obtains an IP connectivity service (e.g., access to the Internet). In this case, the interworking requirements include support of 3G-based access control, signaling between the WLAN and the 3G network for Authentication, Authorization and Accounting (AAA) purposes, etc. Other scenarios can call for more demanding interworking requirements. We may envision, for instance, a scenario in which a 3G subscriber initiates a video session in his home 3G network and subsequently transits to a WLAN environment, wherein the video session is continued seamlessly, i.e., without any noticeable change to the quality of service (QoS). In this case, not only 3G-based access control is required, but also access to 3G-based services is needed over the WLAN network, which in turn calls for appropriate routing enforcement mechanisms. More importantly, however, there is need for QoS consistency across 3G and WLAN, which appears to be not very straightforward given the different QoS features offered by these networks. Indeed, WLANs have initially been specified without paying much attention to QoS aspects and aiming primarily to simple and cost-effective designs. Even with the recent IEEE 802.11e developments, WLAN QoS still exhibits several deficiencies with respect to the 3G QoS. On the contrary, 3G cellular networks were built with the multimedia applications in mind and trade simplicity and cost for inherently providing enhanced QoS in wide-area environments.

In recent years, my researchers have been studying the challenges of seamless session continuity across UMTS (3G) and WLAN, and also evaluate the conditions and restrictions under which seamless continuity is feasible. The most typical scenario considered, is when a number of UMTS subscribers with ongoing real-time video sessions handover to WLAN. For this scenario, the capability of WLAN to provide seamless session continuity under several policy rules and traffic loads is evaluated. One measure we are particularly interested to quantify is the maximum number of UMTS subscribers (also referred to a UMTS roamers) that can be admitted to the WLAN, subject to maintaining the level of UMTS QoS and respecting the WLAN policies. Although most studies focus primarily of QoS consistency, there are also several other issues addresses that are equally important for enabling seamless multimedia session continuity such as, routing enforcement, access control and differentiation between the traffic of regular WLAN data users and UMTS roamers. Typical results suggest that the WLAN can support seamless continuity of video sessions for only a limited number of UMTS subscribers, which depends on the bandwidth reservations, on the WLAN access parameters and on the QoS requirements of video sessions. The operation of the WLAN scheduler has also been proven of paramount importance and schedulers tailored to video traffic can provide significant performance improvement, e.g. in terms of UMTS roamers that can be admitted to WLAN, each one having a video session in progress.

With no doubt, there are still several challenges to be addressed for enabling the seamless interworking of wireless LAN and UMTS networks. As we discussed in this article, the QoS discrepancies between these networks raise some of these challenges. Although the WLAN QoS capabilities have recently been extended considerably with the introduction of IEEE 802.11e, the WLAN is still incapable to support all the QoS features provided by UMTS (e.g., to dynamically control the MSDU error rate, the residual bit error ratio, unequal error protection, etc). This is partially attributed to the characteristics of WLAN physical layer, which has been kept relatively simple in order to enable

## IEEE COMSOC MMTC E-Letter

low-cost designs. As a result, multimedia transmission over WLANs turns to be not so efficient as compared to the UMTS (but definitely less expensive). Vertical handovers present also another challenge for seamless UMTS/WLAN interworking. Since they are usually implemented as mobile-controlled, hard handovers, they bring up considerable QoS concerns, which severely affect the provision of seamless interworking.

As mentioned above, one vital component for the provision of seamless multimedia session continuity is the QoS consistency across the WLAN and UMTS networks. This is vital indeed because without QoS consistency the multimedia sessions will experience different QoS levels in the two network domains and hence seamless continuity will not be doable. It is unfortunate however, that the UMTS and WLAN specifications were based on rather different set of requirements and they ended up supporting rather different set of QoS features. Consequently, the QoS consistency turns to be a quite challenging issue. To provide more insight on this issue, we discuss below a list of WLAN QoS deficiencies with respect to UMTS QoS. Apparently, when we target multimedia session continuity across UMTS and WLAN, we should carefully take these deficiencies into consideration and understand their impact. The discussion is based on the assumption that the WLAN MAC layer complies with IEEE 802.11 plus the amendments of IEEE 802.11e and the physical layer complies with IEEE 802.11g. For a good introduction to the QoS aspects of IEEE 802.11e the reader is referred to "Analysis of IEEE 802.11e for QoS support in Wireless LANs", IEEE Wireless Communications, no. 6, pp. 40-50, Dec 2003.

1. In a WLAN we cannot support unequal error protection across different media streams. For instance, an AMR payload, an H.263 payload and a HTTP payload will all be subject to the same channel coding (for a given transmission rate) and therefore all media streams will be equally protected against transmission errors, no matter their different bit error rate requirements. On the contrary, in UTRAN different radio transport channels (each with its own channel coding) can be established for streams with different error rate requirements. Thus, error rate can be controlled on a per stream basis.
2. Unequal error protection in a WLAN cannot also be supported between different flows in

the same media stream. For instance, the class A and class B bits of an AMR payload will be equally protected against transmission errors, although class B bits can tolerate higher bit error rate. In fact, the WLAN layers are unable to distinguish the different flows of the AMR stream. On the contrary, UTRAN typically employs different radio transport channels for the different AMR flows.

3. Although different media streams may tolerate different residual bit error rates, in a WLAN there is no way to control the residual bit error rate. This is because the WLAN has been optimized to support data streams and therefore enforces a very small residual bit error rate (by using 32-bit long CRC codes). In practice, this may result in increased MAC Service Data Unit (MSDU) loss rate (especially when acknowledgements are not used) since all erroneous packets will be dropped even if some of them could be tolerated by the application. Moreover, the channel utilization will be decreased because, according to 802.11, when a station receives an erroneous MSDU it cannot access the channel until after an Extended Inter-Frame Space (EIFS) period.
4. WLAN stations cannot have dedicated radio channels as in UTRAN and therefore the queuing delay and jitter figures could be increased. Note that in 802.11e Hybrid Coordinator Channel Access (HCCA) mode the stations transmit with a polling discipline and hence delay and jitter will depend on the overall number of stations requesting resources and on the scheduler characteristics. In 802.11e Enhanced Distributed Channel Access (EDCA) mode the stations transmit with a random access discipline tailored to support several different traffic classes.
5. In the WLAN there is no adaptive mechanism for controlling the MSDU loss rate in real-time. The typical way to control MSDU loss rate is via link adaptation with transmission rate change. However, this adaptation is not mandatory in all WLAN stations, it is implementation dependent and, more importantly, it is typically based on some predefined loss rate thresholds, which do not correlate directly with the loss rate requirements of the transmitted media streams. The IEEE 802.11g standard allows

## IEEE COMSOC MMTC E-Letter

the transmit power level to vary (the same holds true for 802.11b and 802.11a) but in practice all WLAN stations tend to use the maximum power level at all times, since no fast power control mechanism exists. Note also that in 802.11e EDCA loss rate is even harder to control since collisions are unavoidable.

6. There are no soft handovers in WLAN. Handovers are typically hard in nature, i.e., follow the break-then-make approach, and hence considerable transmission disruptions may exist that result in QoS degradation. Moreover, handovers in 802.11 are solely controlled by the WLAN stations, so the WLAN infrastructure cannot provide tight control of the QoS provisioning. If a WLAN station tends to “ping-pong” between two APs the QoS will be severely affected and the WLAN infrastructure has no means to prevent that.

One of the main conclusions is that the “Service Data Unit (SDU) Error Ratio” and the “Residual Bit Error Ratio” attributes used in UMTS QoS profile cannot be negotiated and controlled in an 802.11 WLAN, mainly due to physical layer restrictions. Also, the WLAN infrastructure is nearly impossible to guarantee a strict QoS level given that there is no standardized mechanism for soft handovers. Of course, these deficiencies represent the price we pay for facilitating simple and cost-efficient WLAN designs. Nevertheless, it is important to stress out that the above QoS deficiencies of WLANs do not necessarily mean that seamless session continuity from UMTS to WLAN cannot be supported. Under certain conditions, seamless session continuity can be provided even with inefficient utilization of WLAN radio resources (but these are cheap anyway!).



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