MULTIMEDIA COMMUNICATIONS TECHNICAL COMMITTEE MMTC

http://www.comsoc.org/socstr/org/operation/techcom/multimedia.html

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The MultiMedia communications Technical Committee (MMTC) is a volunteer group that examines systems, applications, services and techniques in which two or more media are used in the same session. These media include, but are not restricted to, voice, video, image, music, data, and executable code. The scope of the committee includes conversational, presentational, and transactional applications and the underlying networking systems to support them.

MULTIMEDIA COMMUNICATIONS

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Joining MMTC is easy. Simply send an email to list@comsoc.org with the following string in the body of the message (NOT the subject line):
join multicomm

The mailing list, multicomm@comsoc.org is the communication channel with the MMTC. To post a message to the list, send e-mail to multicomm@comsoc.org.

You can also navigate through MMTC mailing list archive (since Feb. 2004).
http://barbarian.comsoc.org/comsoc.org/multicomm/

Future MMTC Meetings

GLOBECOM 2006 MMTC Activities
(27 November - 1 December 2006)
Multimedia Communications Symposium

MMTC OFFICERS

CHAIR
Nelson Fonseca
Instituto de Computação
Universidade Estadual de Campinas (UNICAMP)
Caixa Postal 6176
13084-971 Campinas, SP
Brazil
Phone: +55 19-3788-5878,
Fax: +55 19-3788-5847
nfonseca@dcc.unicamp.br

FORMER CHAIR
Heather Yu
Panasonic Digital Networking Laboratory
2 Research Way, 08540
Princeton, NJ, USA
Phone: +1 609-734-7325
Fax: +1 609-987-8827
heathery@research.panasonic.com

VICE CHAIRS
North America
Giovanni Pau
Department of Computer Science
UCLA
3803A Boelter Hall
420 Westwood Plaza
Los Angeles, CA, USA 90095-1596
Phone: +1 310-206-3212
Fax: +1 310-825-7578
gpau@cs.ucla.edu

Europe/Africa
Marco Roccetti
Dipartimento di Scienze dell'Informazione
Università di Bologna
Mura Anteo Zamboni 7, 40127
Bologna, Italy
Phone: +39 051–209-4503
Fax: +39 051-209-4510
roccetti@cs.unibo.it

Asia/Pacific
Qian Zhang
Department of Computer Science
Hong Kong University of Science and Technology
Clear Water Bay, Kowloon
Hong Kong
Phone: + 852 2358-8766
qianzh@cs.ust.hk

SECRETARY
K.P. Subbalakshmi
Department of Electrical and Computer Engineering
Stevens Institute of Technology
208 Burchard Building
Hoboken, NJ 07030
Phone: +1 201-216-8641
Fax: +1 201-216-8246
ksubbala@stevens-tech.edu

FORMER VICE CHAIRS

Pascal Frossard
École Polytechnique Fédérale de Lausanne (EPFL)
1015 Lausanne, Switzerland
Phone: +41-21-693-5655
Fax: +41-21-693-7600
pascal.frossard@epfl.ch

Yucel Altunbasak
Georgia Institute of Technology
Atlanta, GA, USA 30332-0250
Phone: +1 404-385-1341
Fax: +1 404-894-8363
yucel@ece.gatech.edu

Gary Chan
Department of Computer Science
Hong Kong University of Science and Technology
Clear Water Bay, Kowloon
Hong Kong
Phone: +852-2358-6990
Fax: +852-2358-1477
gchan@cs.ust.hk

FORMER SECRETARY
Giovanni Pau
Department of Computer Science
UCLA
3803A Boelter Hall
420 Westwood Plaza
Los Angeles, CA, USA 90095-1596
Phone: +1 310-206-3212
Fax: +1 310-825-7578
gpau@cs.ucla.edu

A WORD FROM THE E-LETTER E-I-C

Marco Roccetti

The goal of the E-Letter is to disseminate issues that focus on opinions, initiatives, scientific achievements and perspectives of multimedia with an emphasis on the communication technologies. The current issue of the E-Letter features a report from a conference and a perspective article.

The report is provided by J.C. De Martin on the first Multimedia Communications Workshop, MULTICOMM, that took place at ICC 2006.

The perspective article is provided by G. Marfia and K. Duy Nguyen from the Computer Science Department of UCLA. In their article, titled "Rate Adaptation using SIP", the authors present a service initiation and migration system for real-time communications in a ubiquitous networking environment.

Most important to mention, the complete list of the new committee officers is reported on page 2. We wish them a proficient work while we express our gratitude to the former officers for their hard work and great results achieved!

We renew the invitation to everyone to become regular contributor by submitting proposals for columns, perspective articles and annotated bibliographies. Information for submissions can be found at the MMTC website:
http://www.comsoc.org/~mmc.

Enjoy this issue!

Marco Roccetti
Editor-in-Chief

COSPONSORING / RELATED CONFERENCES AND WORKSHOPS

GLOBECOM 2006

November 27 - December 1, 2006
San Francisco, California, USA
The objective of this conference is to provide a platform for researchers and technologists to present new ideas and contributions in the form of technical papers, panel discussions, as well as, test-bed implementations and real-world evaluation of many ideas in wireless communications. IEEE Globecom 2006 will feature also a Multimedia Communications Symposium.

CCNC 2007

January 11-13, 2007
Las Vegas, Nevada, USA
IEEE Consumer Communications and Networking Conference (CCNC) will present the latest approaches and technical solutions in the areas of consumer networking, enabling technologies such as middleware and multimedia, and novel applications and services. CCNC 2007 will include a peer-reviewed program of technical sessions, technology application panels, tutorials, and poster/demo sessions.

ICC 2007

June 24-28, 2007
Glasgow, Scotland, UK
The IEEE International Conference on Communications (ICC 2007) will be held in Glasgow, Scotland, from 24-28 June 2007. The Conference is aimed at addressing key themes on "Smart Communications Technologies for Tomorrow". The programme will feature a General Conference, ten Specific Symposia, Applications Sessions and Tutorials. Prospective authors are invited to submit original technical papers for oral or poster presentations at ICC 2007 and publication in the Conference Proceedings.

ICME 2007

July 2-5, 2007
Beijing, China
IEEE International Conference on Multimedia & Expo is a major annual international conference with the objective of bringing together researchers, developers, and practitioners from academia and industry working in all areas of multimedia. ICME serves as a forum for the dissemination of state-of-the-art research, development, and implementations of multimedia systems, technologies and applications. ICME is co-sponsored by four IEEE societies including the Circuits and Systems Society, the Communications Society, the Computer Society, and the Signal Processing Society. The conference will feature world-class plenary speakers, exhibits, special sessions, tutorials, and paper presentations.

CONFERENCE CALENDAR

CONFERENCE	LOCATION	INFORMATION
EntNet 06 International Conference on Enterprise Networking & Services	September 11-13, 2006 Vancouver, British Columbia, Canada	http://www.ieee-entnet.org/2006/
PIMRC 06 IEEE Symposium on Personal, Indoor, and Mobile Radio Communications	September 11-14, 2006 Helsinki, Finland	http://www.pimrc2006,org/
MobiMedia 06 Mobile Multimedia Communications Conference	September 18-20, 2006 Alghero, Sardinia, Italy	http://www.mobimedia.org/
MILCOM 06 IEEE/AFCEA Military Communications Conference	October 23-25, 2006 Washington, DC, USA	http://www.milcom.org/
GLOBECOM 06 IEEE Global Telecommunications Conference	November 27 - December 1, 2006 San Francisco, CA USA	http://www.ieee-globecom.org/2006/
ISM 07 IEEE International Symposium on Multimedia	December 11-13, 2006 San Diego, CA USA	http://ism2006.eecs.uci.edu/
CCNC 07 IEEE Consumer Communications and Networking Conference	January 11-13, 2007 Las Vegas, NV USA	http://www.ieee-ccnc.org/
DRM'07 IEEE International Workshop on Digital Rights Management Impact on Consumer Communications	January 11-13, 2007 Las Vegas, NV USA	http://www.ieee-ccnc.org/callforpapers/DMR_workshop/index.html
HWN-RMQ'07 IEEE International Workshop on Heterogeneous Wireless Networks: Resource Management and QoS	January 11-13, 2007 Las Vegas, NV USA	http://www.ieee-ccnc.org/callforpapers/HWN-RMQ_workshop/index.html
NIME'07 International Workshop on Networking Issues in Multimedia Entertainment	January 11-13, 2007 Las Vegas, NV USA	http://www.ieee-ccnc.org/callforpapers/NIME_workshop/index.html
CRN'07 Workshop on Cognitive Radio Networks	January 11-13, 2007 Las Vegas, NV USA	http://www.ieee-ccnc.org/callforpapers/CRN_workshop/index.html
P2PM'07 Workshop on Peer-to-Peer Multicasting	January 11-13, 2007 Las Vegas, NV USA	http://www.ieee-ccnc.org/callforpapers/P2PM_workshop/index.html
ICC 07 IEEE International Conference on Communications	June 24-28, 2007 Glasgow, Scotland, UK	http://www.ieee-icc.org/2007/
ICME 07 IEEE International Conference on Multimedia and Expo	July 2-5, 2007 Beijing, China	http://research.microsoft.com/conferences/ICME07/

MMTC INTEREST GROUPS

Based on the research interests of MMTC members, several IGs have been initiated led by experts and active researchers in each area. Detailed info about the IG charters, focus areas of each IG, and their activities are announced at
http://www.comsoc.org/~mmc/
and through the reflector. The IGs are:

(MSIG) Media Streaming
Chair: Pascal Frossard
Vice-chair: Juan Carlos de Martin

(HNIG) Home Networking
Chair: Prof. Madjid Merabti
Vice-chair: Heather Yu

(MobIG) Mobile and Wireless Multimedia
Chair: Prof. R. Chandramouli
Vice-chair: Oliver Wu

(SecIG) Multimedia Security
Chair: Suba Subbalakshmi
Vice-chair: Deepa Kundur

(QoSIG) Quality of Service
Chair: Qian Zhang
Vice-chair: Apostolis Salkintzis

(ACIG) Interest Group on Autonomic Communications
Chair: Xiaoyuan Gu
Vice-chair: Jiang (Linda) Xie

Call for New IG Proposal, Chair Nominations and Volunteers, and Members

New IG Proposal: The purpose of launching the IGs within MMC is to foster a better community, to get more MMC members involved in our activities and to provide more opportunities to our members. Therefore, we shall support the initiation of new IGs when enough interests are shown. Proposals of new IGs are highly encouraged. Proposals should be sent to the MMC chair via email. Discussion of new IG proposal with MMC officers is also encouraged.

IG Chairs Nomination and Volunteers: We encourage you to volunteer for the available positions. It is a great networking opportunity. Furthermore, it gives you new means to contribute to the technical activities and to promote your career in multimedia communications area. Nomination and volunteers should be sent to the MMC chair via email.

IG Membership: Membership is free. Information about how to join each IG will also be available at each IG will be available at the MMTC Web site. Please stay tuned.

Interest Group on Autonomic Communications

A new IG has been approved, named ACIG, i.e., Autonomic Communications Interest Group.
IEEE ACIG Membership gives you the opportunity
- to network with technical experts in Autonomic Communications,
- to contribute to the technical activities in Autonomic Communications.

Joining IEEE ACIG is free and easy. Simply go to the membership subscription page at:
https://www.ibr.cs.tu-bs.de/cgi-bin/mailman/listinfo/ieeeacig

The mailing list, ieeeacig@ibr.cs.tu-bs.de is the communication channel with the ACIG. To post a message to the list, send e-mail to
ieeeacig@ibr.cs.tu-bs.de.

The mail archives are located at:
http://www.ibr.cs.tu-bs.de/pipermail/ieeeacig

AWARDS

MMC is rolling out two committee awards given to our outstanding members to encourage and promote research and services in the multimedia communications technical areas.

Award Recipients for 2004

Best Paper Award:
Wuttipong Kumwilaisak, Y. Thomas Hou, Qian Zhang, Wenwu Zhu, C.-C. Jay Kuo, and Ya-Qin Zhang, "A cross-layer quality-of-service mapping architecture for video delivery in wireless networks", published in IEEE JSAC, Dec. 2003

Distinguished Service Award:
Dr. Charles N. Judice, for his exemplary service to the Multimedia Communications Technical Committee and the multimedia communications community at large

Nomination Criteria

MMC Best Paper Award
IEEE ComSoc Multimedia Communications Technical Committee will give a yearly award to the Best Paper in the multimedia communications area. Any paper published in an IEEE ComSoc journal/magazine or in the proceedings of an IEEE ComSoc-sponsored conference, workshop, symposium, in the two years preceding the election, is eligible. The prize is an IEEE plaque signed by ComSoc President.
Nominations are solicited for the Best Paper Award 2005. Papers published in 2003 and 2004 will be considered. Paper nominations have to be sent by email to MMTCawdcommittee@netscape.net, with subject line 'MMC-BPA Nomination'. The nomination should include the complete reference of the paper, author information, a brief supporting statement (maximum one page), the name of the nominator, and an electronic copy of the paper when possible. The hard deadline for paper nomination is set to Sep. 30th, 2005.
An independent subcommittee has been created to evaluate nominated papers, and the Best Paper Award 2005 will be presented at ICC 2006, by the MMC chair, to one of the authors of the best paper. The authors will be notified at least 6 weeks prior to the conference.

MMC Distinguished Service Award
Nominations are accepted until August 1, 2005 for the Distinguished Service Award 2005. The prize is a certificate and an IEEE plaque signed by ComSoc President. The Distinguished Service Award will be given at our yearly MMC meeting during GLOBECOM 2005.

Basis for Judging:
Exemplary service to MMC over a sustained period of time

Eligibility:
- The nominee must be a MMC member at the time of nomination
- The nominee must have been a MMC member for a sustained period of time

Nominations should be sent to MMC Chair using the following format:
- Award name
- Nominator name, affiliation, and contact info
- Nominee name & affiliation
- Up to one page supporting statement

Nominee Solicitation and Selection Process: Nominations shall be solicited by the MMC award committee. Final selection shall be made by the award committee and approved by MMC chair.

MMC Award Subcommittee:
Subcommittee chair and members shall be elected at MMC bi-yearly election meeting. Only the award subcommittee shall make decisions regarding the award.

Chair:
Pascal Frossard

Members:
O. Wu, University of Florida
K.P. Subbalakshmi, Stevens Institute of Technology
M. Roccetti, University of Bologna
C. Wen Chen, Florida Institute of Technology
O. Verscheure, IBM Research, Yorktown Heights
Z. He, University of Missouri Columbia

E-LETTER EDITORIAL BOARD

To all MMTC members: If your postal address, telephone or fax numbers have changed, please update them with the committee secretary. You can review our current records on our web page at http://www.comsoc.org/~mmc/.
If you like to join MMTC Mailing List, the indications how to subscribe/unsubscribe are reported at http://www.comsoc.org/~mmc/membership.html.

IEEE Fellow Nominations Subcommittee

MMC TC has established a new subcommittee for IEEE Fellow nominations.

Chair
Prof. Chang Wen Chen

Committee Members
Prof. Charlie Judice
Prof. Homer Chen

Guidelines

1. Working with ComSoc Fellow Committee to identify and evaluate worthy candidates.
2. Promote worthy MMC members to enhance their profiles within ComSoc as well as IEEE institute wide to get ready for Fellow nomination. These will include nominating worthy candidates to society level offices and society level conference organizations.
3. Help with individual MMC members in their Fellow nomination process with advices on how to prepare a strong nomination.
4. Prepare endorsement letters for MMC members when they are nominated for Fellow election.

REPORT FROM A CONFERENCE

J.C. De Martin

Juan Carlos De Martin
IEIIT-CNR, Politecnico di Torino
C.so Duca degli Abruzzi 24,
10129 Torino, Italy
demartin@polito.it

The TC-sponsored First Multimedia Communications Workshop, MULTICOMM, took place at ICC 2006 in Istanbul on Sunday, June 11th.

The workshop featured a keynote speech (Prof. Levent Onural from Bilkent University) on 3DTV, eight oral presentations (62% paper acceptance ratio) and a final discussion panel that produced - with the audience's participation (more than 20 people) - a brief document on the future of multimedia communications.

The workshop has been characterized by active involvement of the participants (students as well as well-established experts) and by a cross-layer mindset, that produced its arguably most interesting results during the final panel, when experts from the physical to the application layers shared their diverse experience and points of view.

The Workshop website (http://multicomm.org) hosts the papers, the presentations and the final document. A stream-on-demand service is also available on the web site. You will find the keynote address audio recording and the audio-video panel discussion.

Regarding the proceedings, the authors agreed to publish their works under the Creative Commons Attribution-NonCommercial-NoDerivs 2.5 License (see http://creativecommons.org/licenses/by-nc-nd/2.5/). As you can see from the link, the license authorizes the copy and dissemination of the work under three conditions: that the authors are always credited, no commercial use, and the work must remain as is, i.e., no modifications are allowed. The Creative Commons licenses are widely used not only for music, videos, images, etc., but increasingly also for scientific works, educational material and peer-reviewed journals. See, for instance, the publications of the Public Library of Science, www.plos.org, MIT's OpenCourseWare Project, or Rice's Conexions Project. On this topic, see also the Science Commons web site, www.sciencecommons.org. MULTICOMM thus earned the distinction of being, as far as we know, the first scientific conference whose proceedings are released under a Creative Commons license.

The organizers were pleased to serve the IEEE Multimedia Communications Technical Committee by organizing this small, yet successful ICC workshop, and they would like to thank all the colleagues who helped or simply encouraged them.

Juan Carlos De Martin
on behalf of MULTICOMM Organizing Committee

Rate Adaptation using SIP

Gustavo Marfia, Khanh Duy Nguyen

Computer Science Department
University of California, Los Angeles
Los Angeles, CA 90095-1596
{gmarfia, dadndk74}@cs.ucla.edu

Abstract—**In this paper we present a service initiation and migration system for real-time communications in a ubiquitous networking environment. The key ideas are (1) an auto-rate fallback style (ARF) rate adaptation scheme using SIP for enhanced multimedia delivery (2) CapProbe-based trigger for media adaptation. Ours is a novel approach that combines the use of the SIP protocol, widely recognized protocol for session control, and a capacity estimation tool as CapProbe. Using CapProbe we are able to perform an accurate estimation of the capacity of the link and to switch media codecs when effectively needed. Therefore users can enjoy real-time communications even when they face resource heterogeneity in the devices participating in the communications. The implementation of the system and the experimental evaluation are also described, which demonstrate the users can switch without terminating the session. A loose QoS approach by switching between media codecs has been already tried as described in the [6] website. Furthermore, problems regarding the use of SIP in handoffs and on MANETs have been addressed in [3], [7], and [4].**

Index Terms—**CapProbe, SIP, rate adaptation, MANET.**

I. INTRODUCTION

In this paper, we use SIP to negotiate the rate of multimedia transfer between sender and receiver based on the bandwidth capacity estimation. The sender will constantly probe the bandwidth capacity in the background using CapProbe. When the capacity will go over or under predefined thresholds this will trigger a change of codec scheme. The capacity monitoring process will notify the SIP Communicator (i.e. the SIP Client) to change the codec scheme to adapt to a new bandwidth capacity. The whole process is automated at the application layer, thus there is no user interaction required to adapt the rate.

The rest of the paper is organized as follows. In section II we explain the background of the project, mainly CapProbe and SIP. In section III the basic operations of the protocol are described. In section IV we give a brief description of the JAIN SIP API, reference library for the SIP protocol. In V we describe the implementation and integration of the test-bed. In section VI we describe the experiments performed on the test-bed. Sections VII and VIII conclude this work with the future work and final comments.

II. BACKGROUND

A. CapProbe

In this section we give a brief description of CapProbe, an accurate link capacity estimation tool.

CapProbe uses *packet pairs*. As the name suggests packet pairs is a pair of back-to-back packets that are sent over any network path to estimate the path characteristics. The basic packet pair relies on the fact that if two packets are sent back-to-back and are queued one after another at the narrow link, they will exit the link with a *dispersion* T given by $T=L/B$ where L is the size of the second packet and B the bandwidth of the narrow link i.e., the capacity limiting link. If the two packets have the same size, their transmission delays are the same. This means that after the narrow link, a dispersion of T will be maintained between the packets even if faster links are traversed downstream of the narrow link. Suppose we have a network configuration where node S is the source, node D is the destination, and the link between nodes A and B is the narrow link. The narrow link capacity can then be calculated as: $capacity=L/T$. The packet pair algorithm assumes that the packets will queue next to each other at the narrow link. The presence of cross-traffic can invalidate this assumption.

The main idea underlying CapProbe is that at least one of the two probing packets must have queued if the dispersion at the destination has been distorted from that corresponding to the narrow link capacity. This means that for samples that estimate an incorrect value of capacity, the sum of the delays of the packet pair packets, which we call the delay sum, includes cross-traffic induced queuing delay. This delay sum will be larger than the minimum delay sum, which is the delay sum of a sample in which none of the packets suffer cross-traffic induced queuing. The dispersion of such a packet pair sample is not distorted by cross-traffic and will reflect the correct capacity. Based on this observation, CapProbe calculates delay sums of all packet pair samples and uses the dispersion of the sample with the minimum delay sum to estimate the narrow link capacity.

B. SIP

The main purpose of SIP is to establish sessions between two user agents. A user agent is composed by a client part that sends SIP requests and a server part that accepts requests. SIP works together with the Session Description Protocol (SDP) which is used to describe the session. We here extract from RFCs 3421 and 2327 the basic information needed to understand the protocol behavior, for more details on SIP and SDP please refer to the RFCs from the IETF website.

```
INVITE sip:  esumner @dynamicsoft.com SIP/2.0
From: J. Rosenberg <sip:    jdrosen @dynamicsoft.com>
Subject: That pay increase
To: Eric Sumner <sip:    esumner @dynamicsoft.com>
Via: SIP/2.0/UDP pc13.dynamicsoft.com
Call -ID: 1997234505.56.78@122.3.44.12
Content -type: application/   sdp
CSeq : 4711 INVITE
Content -Length: 187

v=0
o=jdrosen  53655765 2353687637 IN IP4 122.3.44.12
s=
e=jdrosen @dynamicsoft.com
c=IN IP4 122.3.44.12
t=0 0
m=audio 3456 RTP/AVP 0
```

Figure 1: Example SIP INVITE with SDP message as body

SIP Session Establishment. SIP consists of requests and responses. The establishment of a session using SIP consists of an INVITE transaction and an ACK transaction. The calling party sends the SIP INVITE and the callee replies with a 200 OK response. The INVITE message contains the media type and other session details. In Fig. 1 we can see an example of a SIP INVITE message. The rules are based on those of the HTTP protocol. By analyzing the protocol headers we can see that with the SIP INVITE message in Fig. 1 a call is initiated by J. Rosenberg and is directed to Eric Summer. We can also see what type of media will be exchanged in this communication, in particular this is a phone call (easily seen from the "m=..." line in the SDP content).

The role of the response message is to show the agreement to the chosen media type. The final step is an ACK sent by the caller. Hence it is similar to the 3-way handshake of the TCP connection, except that it is happening at the application layer.

Fig. 2 shows an example of a complete exchange of SIP messages that leads to the initiation of a media stream between the two parties.

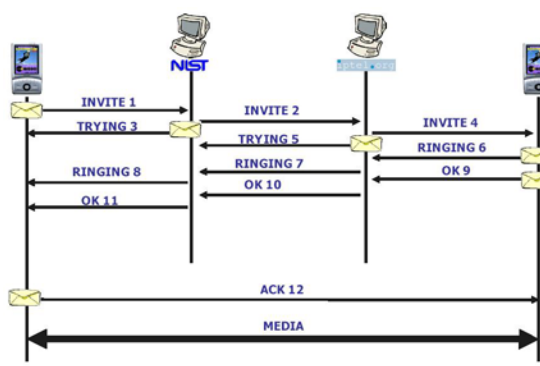

Figure 2: Example of SIP session setup

SIP Reliability mechanisms. To ensure the reliable delivery of the SIP requests and responses involved in the INVITE and ACK transactions, retransmission mechanisms are needed at the User Agent Client (i.e. UAC) and the User Agent Server (i.e. UAS).

1. The client side: The client-side transaction consists of the UAC sending the INVITE request and receiving the 200 OK response. The UAC is aware of the successful transmission of the INVITE request as soon as it receives the 200 OK response. If SIP messages are carried over UDP, the UAC retransmits the INVITE request after an interval that lasts Tr(1) seconds and doubles after each retransmission. The retransmissions cease upon the reception of a 200 OK response at the UAC or after 7 transmissions of the INVITE request. TCP handles packet losses by setting a timer when it sends data and if the data is not acknowledged when the timer expires, it retransmits the data [4]. But for any

type of transport protocol the retransmissions of requests cease when the timer reaches $26 * Tr(1)$ seconds.

2. The server side: The server-side transaction consists of the UAS sending the 200 OK response and receiving an ACK. The UAS is aware of the successful transmission of the 200 OK and at the beginning of the call when it receives the ACK. The retransmission mechanism is identical to the one on the client side. In addition, each 200 OK being received at the UAC triggers the retransmission of an ACK.

III. BASIC PROTOCOL OPERATION

In this section we describe which are the steps involved in adapting to the estimated capacity change, using the SDP and the SIP protocol. As we will see in the following, SIP acts as control channel for the transport of the SDP information. The SDP information is then used at the application level to perform the requested adaptation.

1) SDP (Session Description Protocol) is the payload of SIP messages. By the use of SDP an end-to-end agreement is done on the media format, the ports that will be used and the IP addresses that will be involved in the communication. The session description protocol is a flexible protocol able to describe any media exchange that may take place in a SIP session. The syntax is very simple, we can see in Fig. 1 that a single line is necessary to define the set of media codecs that are supported for a voice communication.

2) While the SIP client runs and a communication has been setup (i.e. the SIP INVITE, 200 OK and ACK messages have been exchanged and the bidirectional audio streams activated by the two parties), the capacity monitoring process runs CapProbe in the background and gets the bandwidth capacity as output of CapProbe. In this version of the software we did not plan to implement passive monitoring, but we planned to perform active monitoring sending dummy probe packets along the path. The process retains the value for predefined number of probing times. If the bandwidth capacity is stable for those times, and the probed capacity reaches certain thresholds for codec schemes, the process will then notify the SIP client to change the codec scheme. The effect will be that the UAC SIP client will construct and send to the UAS SIP client a new SIP INVITE request. In this message, the SDP protocol conveys the information regarding the new codec scheme the UAC wants to use. Upon receiving the new SIP message, the UAS will adjust appropriately its decoder and will start sending its stream encoded as requested by the UAC, if the codec is supported. All this is implemented by using the standard features of SIP. These features are the same that are used when a voice communication is setup and one of the two parties wants to add a video communication. The only difference is that in this case we force a change in the audio codec by making a new request, within the same session, which presents a new media line, for audio, in the SDP payload.

The above points introduce which are the problems that must be faced when implementing such system.

A first problem is that of ensuring that the active probing mechanism doesn't bother the audio streams. CapProbe provides the best available capacity estimation algorithm in terms of accuracy and speed of convergence. The best possible scenario is that represented by an implementation where the CapProbe algorithm is implemented on the top of the media stream exchange. To do this it is necessary to force packet pairs to be sent when sending RTP packets and RTCP control packets.

A second problem is that of finding techniques such that the application layer switch may be performed without degradations in terms the user perceived media quality. This is in fact a very interesting problem, that must also be faced in other scenarios where application layer switching is performed. A well known example is the similar problem that has been faced in the IPTV platforms. A difference with the IPTV case, here, is that since the same voice content is carried by the media streams (i.e. in IPTV we have a change of content when switching from one channel to another, in voice codec switching we do not have a modification of content in the general case), this can be performed in a more loose way. The caveat is that if the channel is experiencing a strong reduction of bandwidth, a slow switch may be perceived as a degradation of voice quality.

A third problem is that of choosing a capacity threshold such that codec switching is triggered. The reader might be wondering why we are concerned about a change in capacity. When taking into account Mobile Ad Hoc Networks (i.e. MANETs) it is easy the case that capacity changes. We are then targeting our design/implementation to such scenario. We don't want the capacity threshold level to be too close to the typical capacity available on the channel since the unavoidable noise in estimates may induce unwanted codec switches. On the other side we don't want the capacity threshold to be too loose, we may experience poor performances by using a codec that requires more bandwidth than the available capacity (this in a general case, but we must be more precise, in the case of audio media this rarely happens, we may then think about the problem in terms of bandwidth saving more than audio quality).

A fourth and final problem is to understand, when the system works, if it is competitive in comparison to other mechanisms that are already available on the *"market"* (i.e. RTP/RTCP for instance). We must then identify if there are and which are the advantages that could be provided by such design. We are here saying that this is an interesting option for such systems that must/want to perform an application layer control on the media. We have already mentioned the IPTV example.

We have here pointed out which are the main issues that may be encountered when designing a system capable to perform application layer codec switching based on capacity estimation information. The system that we have designed takes into account these problems and provides a good starting point to perform various types of experiments.

IV. JAIN SIP

The JAIN APIs are being specified as a community extension to the Java platform, by providing a new level of abstraction and associated Java interfaces for service creation across circuit switched and packet networks (for more information please refer to the Java and Sun websites). The SIP APIs that are of interest and that have either been developed or that are under development within the JAIN initiative are:
(a) JAIN SIP - JAIN SIP is a low level API that maps directly to RFC published by the IETF. These are the SIP APIs that have been used within the project.
(b) JAIN SIP Lite - The JAIN SIP Lite API is a high-level API. The goal of this high-level API is to allow application developers to create applications that have SIP as their underlying protocol without having to have an extensive knowledge of the SIP protocol. This will allow developers to rapidly create applications, such as user agent type applications. JAIN SIP Lite is a thin Java API that can be used as a high-level wrapper around the SIP protocol that will provide application developers with an API that is easy to use.

V. PROTOTYPE IMPLEMENTATION

The software used as starting point for the project comes from NIST (National Institue of Standards and Technologies).

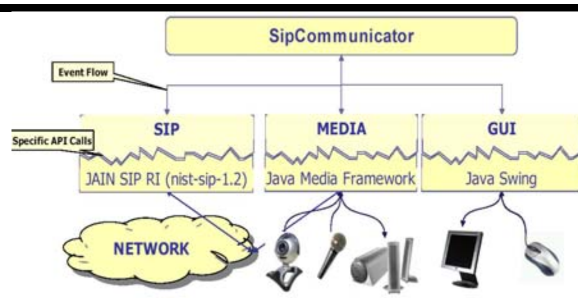

Figure 3: Architecture of the SIP Communicator

The test-bed that has been setup is composed of one Proxy/Registrar server and two clients. We performed our modifications on the clients (i.e. SIP Communicator), while the Proxy and Registrar servers remained unchanged. The clients and the server are all connected with each other by wireless links, using the 802.11b Ad-Hoc mode. The Registrar behaves as a location server, the clients must send a SIP REGISTER request to the Registrar in order to be registered in the SIP network. The Proxy server is the first contact point for each client and, interacting with the Registrar, proxies the requests to the clients.

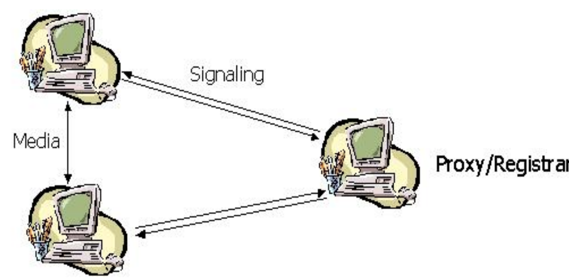

Figure 4: Test-bed

The Proxy/Registrar has been installed on a Linux 2.4 machine. We instead installed the SIP Communicator clients on Microsoft Windows machines. The implementation has been performed by modifying the client in its SIP and media components and by integrating the CapProbe mechanism in the client. The SIP component of the client has been modified allowing it to receive and instantiate requests to change the parameters of a session. In this context we identify the parameters of a session by the media codec that is used to exchange data (audio or video). As specified by the RFC 3261, in order to accept or initiate a new request within the same session, the request must have the same Call-ID header as the first message that initiated the session. The media component of the client, that relies on the JMF libraries (Java Media Framework), has also been modified. This has been the work that required more time since several problems have been encountered to change the media codec, once one had been used. Using a cloning and re-encoding mechanism of the streams this has been finally achieved. Since CapProbe is not available for Windows platform, we chose to simulate it in the Windows environment. In this set of experiments we were interested to evaluate the efficiency of the switching mechanism, which we perceived as the most important and interesting problem. In addition, the bandwidth required by audio streams is so small in comparison to the capacity provided by the wireless channel that it would be basically impossible to observe codec switches triggered by capacity threshold traverse. The capacity simulation package is then very simple, a simulation program randomly outputs the random bandwidth capacity. We set the number of consecutive times that the simulation needs to output the bandwidth capacity, in order to interpret it as a capacity estimate. If the case happened, then we select the last output bandwidth capacity as the actual probing value and process on it. If this value is different from the last successful probing value this triggers the change to the codec scheme, the probing process will notify the SIP Communicator to send re-invite message to the callee to change codec.

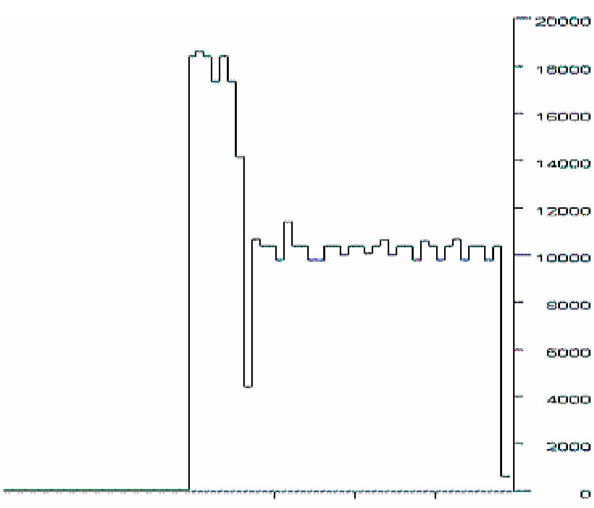

Figure 5: In this image we have a switch of media from the ITU-T G.711 codec to a DVI4 8000 samples/sec codec, the graph is expressed in bytes/sec over seconds.

VI. Experiments

Once the test-bed has been setup and the modified client, integrated with the CapProbe simulator, has been debugged, we ran some experiments. The results we here present are the time to switch from one codec to another, once triggered from CapProbe (in this case the simulator). In condition of no mobility, we got a switching time of about 0.6 seconds, as may be seen from the Fig. 5 and 6. From the figure it is possible to see that there is a peak after a minimum, that is due to the arrival of delayed packets. The codecs we considered are G.711 PCMU, DVI4 8000 samples/s and GSM. The time to switch is the same, from the result of a ten minutes run of the client and approximately 30 observed switches. From the user experience point of view, the switch is almost not perceivable (we must consider that during a voice communication there is the probability that the change may happen during a silent period). We have to point out that 0.6 seconds are anyway noticeable on a voice communication, since the human ear is able to perceive delays that are above the 150-180ms threshold.

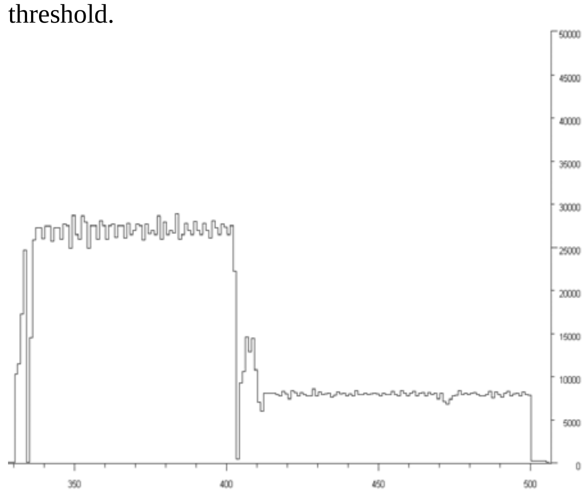

Figure 6: In this image we can appreciate a switch of media from the ITU-T G.711 PCMU codec to the GSM codec, the graph is expressed in bytes/sec over seconds.

Another disadvantage that may be here highlighted is that this system may provide much worst performance if a switch of video codecs was considered, since we have no silence periods in video streams. We again point out that there are already available systems that may perform better in the specific case, but we are here trying to address those systems that require a strict control at the application layer. The use of a buffer at the receiver side improves the performance, but this must be done reminding that an increased buffer at the receiver side might result in the lose of interactivity in the conversation. We anyway prove that a system like this can be well suited to a voice communication in which we want to adapt the load to the current channel conditions, where the channel condition information is fed at the application layer. The other result we got is that we integrated a SIP network in the lab and that this will be available for future experiments and work. In order to run experiments with mobility and get more interesting results we will have to integrate the real CapProbe. In that scenario we will be able to perform experiments changing the packet pair interval and varying the distance/detected capacity.

VII. FUTURE WORK

There are several possible enhancements to the present experimental test-bed.

(a) Port CapProbe to Windows to truly integrate with SIP Communicator;

(b) Perform CapProbe's capacity estimation from the flux of data packets by the client;

(c) Tune CapProbe and the renegotiation timers in presence of different conditions of mobility;

(d) Usage of an available bandwidth estimation tool instead of a capacity estimation tool;

(e) Usage of a more network-aware video transport protocol;

(f) Use the dummynet network simulator to perform extensive testing.

VIII. CONCLUSION

In this paper, we described how the SIP protocol may be used to control a media session and to adapt media coding to channel conditions. Furthermore, we described the issues that may come up by integrating a SIP network with other tools. Our goal is to have a working testbed that integrates media controlled by SIP, CapProbe estimation and Ad Hoc Network routing protocols. Once all this will be setup we will be able to test the adapting mechanism in more interesting scenarios and verify the effective gain against currently working systems.

ACKNOWLEDGMENT

We would like to thank Giovanni Pau and Claudio Palazzi for the interesting discussions related to SIP and its uses.

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CALL FOR CONTRIBUTIONS

Call for Contributions per Annotated Bibliographies for

The Multimedia Communications Technical Committee

E-Letter

Editor in Chief: Marco Roccetti

IEEE Communications Society

The E-letter of the Multimedia Communications Technical Committee of the IEEE Communications Society is an electronic publication that welcomes submissions of annotated bibliographies.

A considerable barrier to entry into a new field of research is to become aware of the existing literature on the topic. The Internet and search engines -such as IEEExplore and, more recently, Google Scholar- have made access conference proceedings and journals immensely easier than it used to be.

However, speed and ease of access, by themselves, do not solve the problem of understanding the state of the art in a given field. Some form of intelligence is needed to filter the raw data represented by the very large number of available publications. Such intelligence may be acquired, in due time, by reading and attending conferences - or it may come from experts already working in the field.

To help fellow engineers and researchers to gain easier access to new fields of activities, the E-Letter of the Multimedia Communications Technical Committee (MMTC) invites multimedia experts to submit annotated bibliographies on topics of their choosing.

It is expected that the annotated bibliographies could be of various kinds – from tutorial level bibliographies on the general field of multimedia communications to bibliographies on very specialized subtopics.

If technically feasible, we will adopt an open approach to bibliographies development. Instruments such as wiki are, in fact, making very easy to build knowledge repositories in a collaborative fashion, as shown, for instance, by the astounding success of wikipedia.org. Initial contributions could, therefore, if the original author agrees, be placed on a MMTC wiki to be integrated by comments and modifications made by the community at large. The E-letter will periodically publish selected annotated bibliographies.

Possible topics for annotated bibliographies include, but are not limited to:

- Hardware and Software for Multimedia
- Home Networking for Multimedia
- Implemented Prototypes
- Mathematical Modeling and Simulation for Multimedia
- Mobile and Wireless multimedia
- Multimedia Communication Systems
- Multimedia Security
- Multimedia Design
- Multimedia Development Tools
- Multimedia Networking and Quality of Service
- Networked Multimedia Entertainment
- Quantitative and Qualitative Studies for Multimedia
- Streaming Multimedia
- Theoretical/Ergonomic Issues Regarding Multimedia Communications

Annotated bibliographies will be subject to peer review and, upon acceptance, published in an upcoming issue of the E-Letter. All authors should consider the general nature of the E-Letter's readers. Annotated bibliographies should not have been previously published and must not be submitted for publication as well.

Submission guidelines are as follows: length should be no more than 3000 words (four double column pages).

Annotated bibliographies should be submitted in pdf format by e-mail to the E-Letter Assistant Editor J.C. De Martin at demartin@polito.it.

Deadlines:
The next issue of the E-Letter will appear on December 2006. Our deadline for receiving annotated bibliographies articles is 60 days prior to the cover date.

CALL FOR CONTRIBUTIONS

Call for Perspective Articles for

The Multimedia Communications Technical Committee

E-Letter

Editor in Chief: Marco Roccetti

IEEE Communications Society

Multimedia technology, networks and services are making productive use of important innovations in technical parallel fields: from signal processing and compression to storage and switching devices; from satellite and fiber -based communications to computer graphics and animation; from mobile and wireless systems to information security. A beneficial aspect of this phenomenon is that it is pulling together an extremely diverse group of experts specializing in technical converging areas. Even though such an ever-evolving environment promotes interdisciplinary fusion, however, teachers, researchers and professionals of the discipline need access to the most current information about the concepts, issues, trends and technologies in this emerging field. The **E-Letter** of the **Multimedia Communications Technical Committee** wishes to become a fast medium that provides a comprehensive coverage of the most important definitions, concepts, issues, trends and technologies in the field of multimedia communications technology. To this aim, the **E-Letter** of the Multimedia Communications Technical Committee welcomes submissions of Perspective Articles. Perspectives are articles written from the point of view of an expert in the multimedia technology field. They should focus on a particular technology or technology-related issue and how that technology or technology-related issue is being implemented and is impacting the multimedia arena. The E-Letter is seeking perspective articles on the subject of multimedia as it applies to the broad spectrum of multimedia communications. Also manuscripts for short essays and opinions may be considered.

Possible topics include, but are not limited to:

- Hardware and Software for Multimedia
- Home Networking for Multimedia
- Implemented Prototypes
- Mathematical Modeling and Simulation for Multimedia
- Mobile and Wireless multimedia
- Multimedia Communication Systems
- Multimedia Security
- Multimedia Design
- Multimedia Development Tools
- Multimedia Networking and Quality of Service
- Networked Multimedia Entertainment
- Quantitative and Qualitative Studies for Multimedia
- Streaming Multimedia
- Theoretical/Ergonomic Issues Regarding Multimedia Communications

Selected articles will be peer-reviewed and, upon acceptance, published in an upcoming issue of the E-Letter. All authors should consider the general nature of *E-Letter's* readership. Manuscripts should not have been previously published and must not be submitted for publication elsewhere. The **basic format to follow** is:

- Introduce the technology or issue being discussed.
- Discuss the technology's current or future impact on multimedia communications.
- Discuss pros and cons of the technology/issue.
- Discuss what the author is doing regarding this technology/issue.

Other Guidelines are as follows:

- Length should be no more than 2,000 words (three double-column pages).
- Articles should contain no more than 3 Figures. Figures and tables count for 300 words.
- Articles must contain no more than six references.
- Articles should be submitted in a .pdf format by e-mail to roccetti@cs.unibo.it.

Deadlines:

The next issue of the E-Letter will appear on December 2006. Perspectives are generally scheduled far in advance. Our deadline for receiving completed articles is 60 days prior to the cover date. We may accept some material later than that, but special arrangements must be made in advance with the Editor.

CALL FOR CONTRIBUTIONS

Call for Columns for

The Multimedia Communications Technical Committee

E-Letter

Editor in Chief: Marco Roccetti

IEEE Communications Society

The **E-Letter** of the **Multimedia Communications Technical Committee** features columns written by recognized experts in all the technological fields related to multimedia communications. Columns should give to all the multimedia community partners a possibility to voice their views on the issues, challenges, and opportunities facing industry and academia in connection with the field of multimedia communications. Columns featured by the E-Letter of the Multimedia Communications Technical Committee are intended to become a fast medium that provides a comprehensive coverage of the most important issues, concepts, definitions, trends and techniques in the field. To this aim, the E-Letter is looking for a group of insightful and diligent volunteers to serve as regular (or sporadic) columnists on the 2004-2005 term. Columns will be considered on all the aspects of multimedia communications. The E-Letter offers an unparalleled opportunity for potential columnists to express thoughts and opinions to a community-wide audience provided that the following instructions are followed.

What does it mean to be a columnist for the E-Letter?

It means keeping informed about multimedia issues, as well as news and scientific headlines. It means thinking about the issues that matter to readers in the context of the multimedia communications community. It means undertaking substantial research. It means writing clearly and effectively (perhaps provocatively) to demonstrate an opinion piece that can be easily followed.

What is a column for the E-Letter?

Columns are very brief articles in form of opinions, short essays, or news written from the point of view of an expert. Even though a column is, in essence, a timely and relevant piece of opinion writing, each good E-Letter column should relate an opinion to the most relevant topics of the multimedia community. Also controversial issues can make for a great column, but only if they sound interesting for the multimedia community.

Who can be a columnist for the E-Letter?

Well known experts, skilled practitioners, professionals and researchers are welcome to submit ideas for E-Letter columns. Also contributions from Chairs or members of the various Interest Groups of the Multimedia Communications Technical Committee, as well as from any member of ComSoc, discussing issues related to the activities of their groups, are greatly appreciated. The real and final qualification is having something interesting to say about multimedia communications and its surrounding community, and a willingness to put in the necessary time and effort.

Selected columns will be evaluated by the E-Letter Editor and, upon approval, published in an upcoming issue of the E-Letter. The basic format to follow is:

- Length should be no more than 700 words in length (one double-column page).
- Columns should contain no Figures.
- Columns should contain no References.
- Columns should be submitted as plain text (ASCII) by e-mail to roccetti@cs.unibo.it.

Deadlines:

The issue of the E-Letter will appear on December 2006. Our deadline for receiving columns is 15 days prior to the cover date. We may accept some material later than that, but special arrangements must be made in advance with the Editor.