

2002 MIPRO

S MIPROM U DRUŠTVO ZNANJA
WITH MIPRO TO KNOWLEDGE SOCIETY



IEEE Region 8

upro

www.mipro.hr



XXV.

JUBILARNI

MEĐUNARODNI SKUP

JUBILEE INTERNATIONAL

CONVENTION



20.-24.5.2002.
Opatija, CROATIA
Grand hotel Adriatic
Kongresni centar
(Convention Centre),
hotel Admiral,
hotel Istra & hotel Ičići



CTE

- COMPUTERS IN TELECOMMUNICATIONS
- RAČUNALA U TELEKOMUNIKACIJAMA

COMPUTERS IN TELECOMMUNICATIONS
RAČUNALA U TELEKOMUNIKACIJAMA

Redakcija

Stjepan Golubić
Branko Mikac
Ivan Škunca
Mihajlo Filiferović
Željko Zekić
Lučijano Raspor
Tedi Viškanić

Odgovorni urednik

Stjepan Golubić

Tehnički urednici

Željko Zekić
Lučijano Raspor
Tedi Viškanić

ISBN 953-6042-87-8

Nakladnik

HRVATSKA UDRUGA ZA MIKROPROCESORSKE, PROCESNE I INFORMACIJSKE
SUSTAVE, MIKROELEKTRONIKU I ELEKTRONIKU - MIPRO HU, 51001 Rijeka, p.p.
303, tel./fax +385 (0)51 423 984, tel. +385 (0)51 216 209, e-mail: mipro@ri.hinet.hr, WWW:
<http://www.mipro.hr>

Tisak

LINIAVERA d.o.o., Rijeka

Svako daljnje umnožavanje i pretisak tekstova iz zbornika u bilo kojem obliku nije dopušten bez suglasnosti Hrvatske udruge MIPRO, jer tekstovi predstavljaju autorske radove u smislu zaštite autorskih prava

SERVICE OPTIMIZATION BY SELF-ADAPTATION IN THE ORIGINATION AND TERMINATION PART D. Jevtić, D. Sablić, K. Čunko	147
FORMAL SPECIFICATION OF ARCHITECTURE MODEL FOR VARIABLE QoS REQUIREMENTS D. Žagar, S. Rimac-Drlje	153
MULTI-AGENT COLLABORATION IN QoS MANAGEMENT K. Tržec	158
QUALITY OF SERVICE MECHANISMS FOR MANAGING MISSION CRITICAL APPLICATIONS IN CORPORATE NETWORKS D. Galinec	164
 MODELS AND DEVELOPMENT METHODS MODELI I RAZVOJNE METODE	
SIMULACIJA PARAMETARA TELEFONSKOG PROMETA U REALNOM VREMENU Z. Čarapina, Z. Avdagić	171
MESURING AND ANALYSING TRAFFIC GENERATED BY WAP APPLICATION D. Vukomanović, I. Horvat, D. A. Dumančić	176
INFLUENCE ON BASIC CALL SET-UP BY INTEGRATION OF PACKET AND CIRCUIT SWITCHED CORE NETWORK D. Huljениć, T. Galinac	180
SERVICE MOBILITY MODELS FOR VIRTUAL HOME ENVIRONMENT: CASE STUDY OF A MOBILE AGENT BASED SERVICE T. Marenić	186
MACHINE LEARNING IN MOBILE NETWORK DESIGN D. Zrno, D. Šimunić	192
IMPROVEMENTS IN EFFICIENCY OF RAY-TRACING MODELS FOR PROPAGATION PREDICTION G. Šimac, D. Šimunić	196
UML AND SDL IN SOFTWARE DEVELOPMENT S. Dešić, D. Gvozdaniović, D. Huljениć	202
AN EXPERIENCE IN USING FORMAL METHOD IN THE SOFTWARE MAINTENANCE PROCESS ANALYSIS Ž. Car, A. Carić	206

Influence on Basic Call Set-up by Integration of Packet and Circuit Switched Core Network

D. Huljenic, T. Galinac

Ericsson Nikola Tesla

Krapinska 45, Zagreb, Croatia

Phone: +38513653125 Fax:+38513653548 E-mail: darko.huljenic@etk.ericsson.se, tihana.galinac@etk.ericsson.se

Abstract – Evolution of existing telecommunication network towards multi-service network implies an evolution from circuit switched core network towards packet switched core network. By introduction of new network architecture it was necessary to provide access of traditional PSTN/ISDN TDM based networks into packet switched core network. This article describes how the introducing of new network architecture affects basic call. Model of basic call in distributed network architecture of the node is analysed. Duration of call set-up in multiple scenarios are measured and compared.

I. INTRODUCTION

In last few years a great expansion of IP traffic is noticed. Because of using telephony networks as dial-in access to Internet these networks (PSTN/ISDN) were reaching their scalability limits and major investments of network operators were needed to accommodate its growth. On the other hand rapidly growing service market requires upgrades of the performance and functionality of the network. The network architecture is currently vertically specialised which means that different applications have their own access, connectivity network and server node for traffic handling what is not promising for common service platform concept.

Different technology analysis and communications with the customers lead proposed solution to multi-service network architecture. The next generation networks are expected to support multimedia, real-time, and non real-time applications with guaranteed and non-guaranteed Quality of Service (QoS). New network architecture is expected to migrate towards multi service network, which will be modular, adaptable and flexible.

The paper is organised as follows: overview of 3G-network evolution is described in Section 2, nodes and signalling in splitted architecture is defined in Section 3. A basic measurement of splitted architecture is elaborated in Section 4, while Section 5 deals with analysis of achieved results. Section 6 concludes the paper.

II. OVERVIEW OF 3G NETWORK EVOLUTION

UMTS was defined in Europe as the third-generation mobile telecommunication system that would replace the current GSM standard and could offer a much more attractive and richer set of services to the users. A widely accepted assumption at the time was that the core network of UMTS would be based on broadband integrated services digital network (B-ISDN), asynchronous transfer mode (ATM) and intelligent network (IN) techniques.

During the time IP traffic has been increasing exponentially and it exceeded telephone traffic. On the other hand mobile traffic is also growing exponentially and it takes one of the most important places in telecommunications. One of the reasons for the success of the Internet is that it enables anyone to create and make available new services and has resulted in a high degree of innovation. So, the next generation of networks has to be more open than traditional PSTN networks, meaning the requirement for network equipment with open and standardised interfaces.

The core network contains circuit-switched and packet-switched domain. Initially, the circuit-switched domain was more important than packet-switched. However, the increasing demand for IP connectivity has reversed these priorities.

Two trends of evolution of next generation of networks are visible. The first one is design of UMTS network architecture moving towards an all-IP network, and the second one is design of the UMTS service architecture with standardised open network interfaces.

Discussions on making standards for third-generation mobile system with core network started in the beginning of 1998. This project was called the Third Generation Partnership Project (3GPP). Simultaneously ITU-T initiated and investigated the service and network capability requirements beyond IMT-2000 in close collaboration with 3GPP.

Since mid-1999 two trends turn into the focus of investigation in all of standardisation projects within UMTS. This implies replacement of circuit-switched transport technologies, which were still used in UMTS first phase, by packet-switched (e.g. IP) transport technologies and introducing of multimedia support in the UMTS core network.

In the all-IP core network, all data are transferred on IP, including even traditional circuit-switched voice and data. Circuit-switched voice is optimised in terms of bandwidth and quality. Packet-switched mode is more flexible in terms of services supported and allows the introduction of multimedia application, but it is less efficient in terms of bandwidth. There are two major protocols for supporting VoIP: SIP, standardised by IETF, and H.323, standardised by the ITU. Recently, it was decided in 3GPP to use only SIP as call control protocol between terminals and mobile network. Interworking with other H.323 terminals (e.g. fixed hosts) will be performed by a dedicated server in the network.

The requirements for an all-IP core network are summarised as follows:

- support of roaming and handover to 2G networks

- support of 3G circuit-switched terminals in full IP core network, providing backward compatibility
- support of new (e.g. IP and multimedia) as well as existing services, such as speech, SMS, and supplementary IN services.

The technology trend today is to split call control and bearer control functionality into different functional entities. The functional entities communicate with each other with a standardised (open) protocol. The functional entities in the architecture proposed by 3GPP in an all-IP UMTS core network architecture shown in figure 1. according to [1] are the following:

- MSC server:** The MSC server controls all calls coming from circuit-switched mobile terminals and terminated calls from a PSTN/ISDN/GSM network to a circuit-switched terminal. The MSC server interacts with the media gateway control function (MSCF) for call to/from the PSTN.
- Call state control function (CSCF):** The CSCF is a SIP server that provides/controls multimedia service for packet-switched (IP) terminals, both mobile and fixed.
- Media Gateway (MG) at the UTRAN side:** The MG transforms VoIP packets into UMTS radio frames. The MG is controlled by the MGCF by means of Media Gateway Control Protocol, H.248.
- MG at the PSTN side:** All calls coming from the PSTN are translated to VoIP calls for transport in the UMTS core network.
- Signalling Gateway, SG:** relays all call-related signalling to/from the PSTN/UTRAN on an IP bearer and sends the signalling data to MGCF.
- MGCF:** The first task of the MGCF is to control the MGs via H.248. Also, MGCF performs translation at the call control signalling level between ISUP signalling, used in the PSTN and SIP signalling, used in the UMTS multimedia domain.
- Home subscriber server (HSS):** The HSS is the extension of the HLR database with the subscribers multimedia profile data.

There are still many actual problems that have to be solved in 3G but as always, there is another question on what would be next and how the 3G will evolve. The answer to this question is not clear yet but in [3] assumption is made that one of defining factors of 4G will be the cooperation of different access technologies, different access networks and different expected QoS which are needed for particular connection.

III. NODES AND SIGNALLING IN SPLITTED ARCHITECTURE

The separation of network functions into distinct layers is a key technique in new open network architectures. It allows each layer to evolve independently, as technology evolves. It also allows different transport technologies, existing and new ones, to be deployed independently without affecting the control or service layers. AXE 10, which dominates in the circuit-switching world, has been continuously developed and improved over last 25 years. It's primary function has always been to allow the transmission transfer of data and information over the telephone network. Now AXE is getting ready for the future and to allow its smooth migration towards new architectures. The new network architecture reuses as much as possible the installed GSM service base and the established N-ISDN network. The road to IP multimedia is divided into four phases as follows:

- basic UMTS functionality, handover between UMTS-GSM and multimedia over Circuit Switch (CS - H.324) are implemented
- architectural split for CS and ATM transport for CS
- IP real time transport for all services, increased capacity, CAMEL phase 3, UMTS positioning
- Implements 3GPP architecture

The main innovation in the new core network architecture is splitting into layers as shown in figure 2.

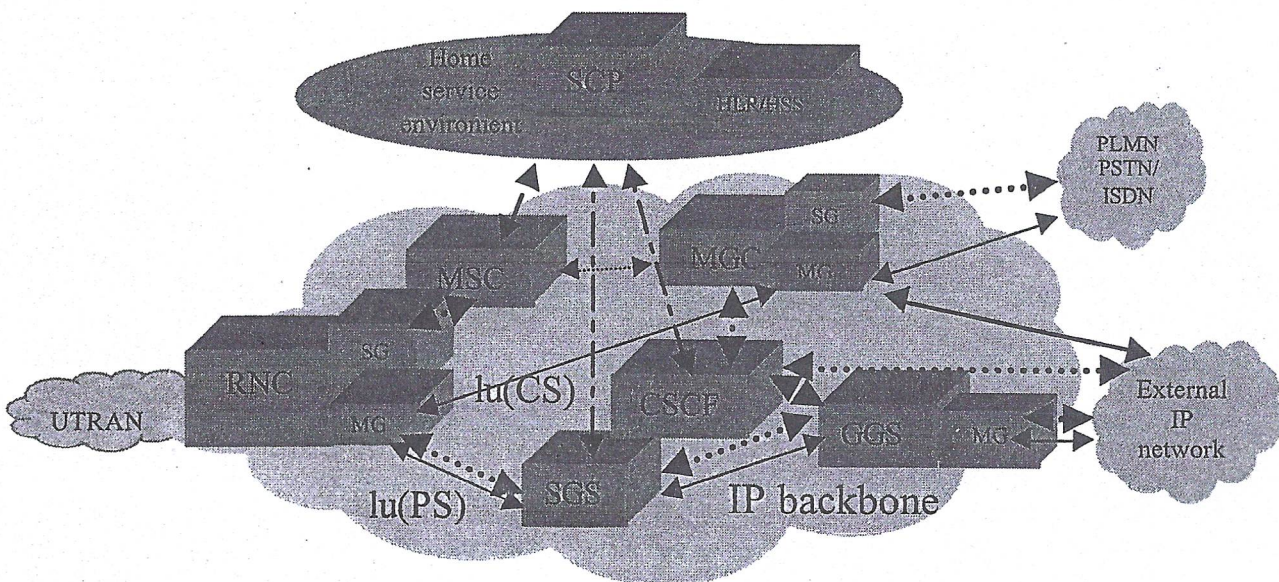


Fig 1. All - IP architecture

The Connectivity layer, at the bottom, contains Media Gateways (MG). MG is a kind of 'transformer' that enables different networks to communicate with each other, making it possible to process packet and circuit switched information in the same network. With MG, operators can easily implement ATM and IP technologies in their networks and make use of the already installed GSM network base. The user plane is transported across the connectivity layer between/via MGs.

The Network Control layer contains Servers and other control nodes. Servers, also called Media Gateway Controllers (MGC), control the MGs using Media Gateway Control Protocol, MGCP (H.248). For communication between Server nodes - Bearer Independent Call Control (BICC) protocol is used. On the top of the Control layer there may be a separate Application layer with Application Servers.

For the signalling within the Core Network, it is assumed that it is based on ATM/IP bearers. That is the reason why a Signalling Gateway is needed to interface external, TDM based networks. The Signalling Gateway performs routing of Signalling System No.7 messages between different types of SS7 signalling data links and includes the Signalling Transfer Point (STP) function.

Introduction of new core network architecture requests to have call control and bearer control protocol independent of each other. The idea is to modify monolithic Signalling System No.7 (SS7) ISUP protocol to serve as a call control protocol. This modified protocol called BICC would offer the full set of PSTN/ISDN services. A whole variety of packet networks (ATM, IP) could serve as the bearer network. The bearer control protocol is used to set up the bearers needed to carry the PSTN/ISDN services. For communication between server nodes in the Network Control layer the BICC protocol is used. The BICC protocol passes information about the selected MG to the succeeding server for bearer establishment. However, in this stage of implementation of next generation network it was necessary to provide access to networks that are external to core network such as PSTN/ISDN networks.

Remote ISUP is the capability of an AXE 10 based server node which is at the edge of core network and interfaces external PSTN/ISDN network to support ISUP protocol using TDM-based bearer network accesses for the ISUP controlled traffic which are located in a remote MG. The purpose of remote ISUP is to allow the implementation of transcoders at the edge of the network. By supporting TDM accesses for remote ISUP in MG, the MGC can control the transcoder in the MG that is used to interface the external ISUP node. If MG did not support this, then only AXE10-based MG could be used for ISUP calls and in order to put the transcoder at the edge, the AXE10-based MG would have to support control by GCP.

IV. MEASUREMENTS

Splitting of architecture lead us to functional benefits but whole concept must be proved from the point of performance. For such purpose it was done a set of different measurements in simulated environment to make a basic sense of gained impacts.

Measurements were performed using SEA (Simulated Environment Architecture) and ISUP traffic simulator (UPSim). SEA provides the possibility to have simulated AXE nodes run on Unix workstation. It is composed of components that simulate different parts of the AXE (e.g. device processors and group switch). In our case UPSim is used to simulate ISUP (ISDN User Part) messages (e.g. IAM) as input trigger to our tested node.

The goal of these measurements was to see the delay brought into call set-up by splitting of node architecture. Measurements are made using the SEA tool, and whole environment is simulated. Because of simulated environment there is no sense to measure delay of the MGCP link and delay inside MG, but this is neither our focus of interest.

We are interested in time loss/gain in AXE node located at the interface between traditional circuit switched core network and packet switched core network.

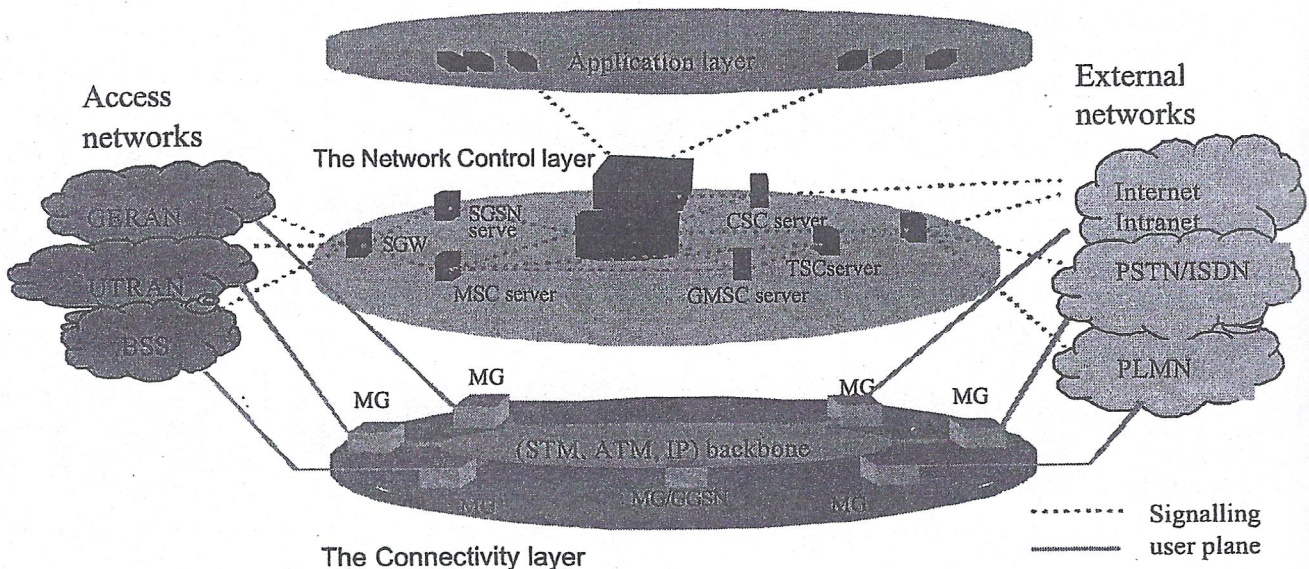


Fig. 2. The main idea of all - IP network, splitting of architecture

We have interest in the AXE node that is transit node toward splitted architecture. For the comparison purposes basic call set-up time is measured also in the case of traditional AXE10 node with collocated MG at incoming and outgoing side, and in case of stand alone node which controls hardware resources located in external MG.

Since time resolution of SEA tool equals 1 ms and the measured times of call set-up in ISUP software complex within AXE node are of the same order of magnitude it is not possible to measure those times precisely. Thus, in some cases measured time was 0 ms meaning that actual time is less than 1 ms. However, the measurements are fine enough to distinguish the difference in call set-up duration in considered cases.

A. Local-Local ISUP call

In this case the time of call set-up within the traditional AXE node in pure circuit-switched network is measured. This means that on incoming and outgoing side of the node the ISUP signalling is used. Call set-up flow when through connection is established within the AXE node is shown in the figure 3. For through connection it is necessary to seize accesses on Group Switch (GS), on incoming and outgoing side. After accesses are successfully seized the bearer is through connected through GS.

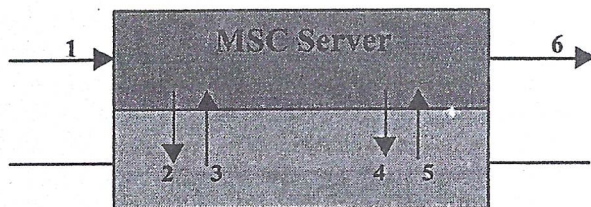


Fig. 3. Local-Local ISUP

Table 1. Results of measuring in Local-Local structure of the node

explanation of measured times	time (ms)
seizing of access on the incoming side	0
seizing of access on the outgoing side	0
through connection	20
complete set-up time in the node	20

B. Local-Remote ISUP call

This is the case that happens when call originates in PSTN/ISDN network and terminates or just passes through core network. The call set-up flow in the node that is located on the access to core network is shown in the figure 4.

Because of splitted architecture the incoming access and outgoing access are in different MGs, but one of the MGs is within the combined node. This implies requests for additional actions for establishing physical connection between combined and external MG.

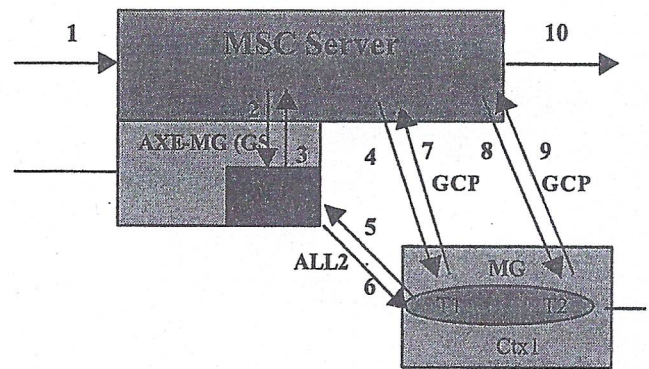


Fig.4 Local-Remote call set-up

When request for establishment has come ISUP Initial Address Message (IAM) to the incoming side in step 1, seizing of access is performed in step 2, figure 4. Access is seized in collocated MG, in step 3. Outgoing side has to seize access too but in external MG (termination T1 in Context 1 - Ctx1), in step 4. When incoming and outgoing access are seized AXE discovers that one access is internally GS-connected and one is connected to another node. It requests set-up of a connection from the external access to the collocated node (internal connection). To be able to perform the connection it has to seize an additional access point in the same context (Ctx1) in external MG and trigger the bearer set-up towards MG, in step 6. When the bearer has been successfully set-up through connection can be performed.

Table 2. Results of measuring in Local-Remote structure of the node

explanation of measured times	time (ms)
seizing of incoming access in collocated MG	2
seizing of outgoing access in external MG	1368
seizing of access for bearer set-up with external MG	790
through connection in GS	20
complete set-up time in the node without time towards external node	22
complete set-up time in the node including interconnection with external node	2180

C. Remote-Local ISUP call

This is the case that happens when call originates or just passes in core network and terminates in PSTN/ISDN network. The call set-up flow in the node that is located on the access to core network is shown in figure 5. Because of splitted architecture the incoming access and outgoing access are in different MGs, but one of the MGs is within the combined node.

When request for establishment has come (IAM), in step 1, to the incoming side which is remote ISUP, seizing of access is performed in step 2, figure 5.

When incoming access is successfully seized then an outgoing side seizes access for remote connection. When incoming and outgoing accesses are seized AXE discovers that one access is internally GS-connected and one is connected to external MG.

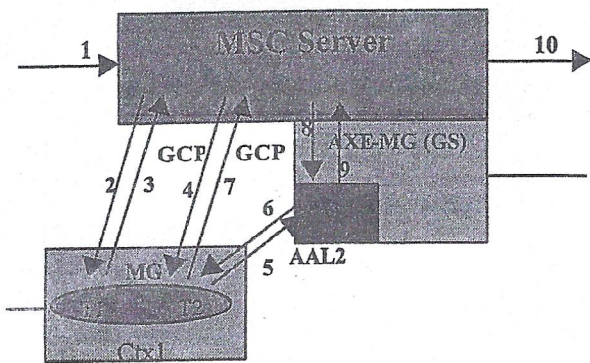


Fig. 5. Remote-Local call set-up

To be able to perform through connection it triggers the bearer set-up towards external MG. When the bearer has been successfully set-up through connection can be performed.

Table 3. Results of measuring in Remote-Local structure of the node

explanation of measured times	time (ms)
seizing of incoming access in collocated MG	820
seizing of outgoing access in external MG	2
seizing of access for bearer set-up with external MG	1198
through connection in GS	20
complete set-up time in the node without time towards external node	22
complete set-up time in the node including interconnection with external node	2040

D. Remote-Remote ISUP call

This is the case that will never probably happen but it can be measured and tested by SEA simulator. Here it is given as an additional case only for comparison purposes. Seizing of TDM accesses in external MG by both sides (incoming and outgoing) does not make sense in core network because of packet switched backbone. For that purposes BICC as a signalling protocol is used.

In this case, when incoming and outgoing access are located in external MG, AXE has no contact with bearer, so it has not to perform physical switching function. AXE has only to perform logical switching and request MG to perform physical switching.

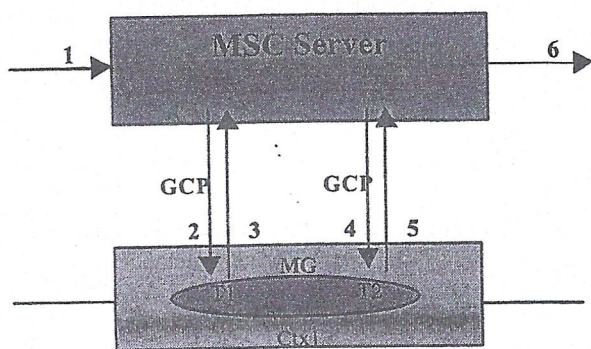


Fig.6. Remote-Remote call set-up

Table 3. Results of measuring in Remote-Remote structure of the node

explanation of measured times	time (ms)
seizing of incoming access in external MG	820
seizing of outgoing access in external MG	988
logical through connection in AXE	2
complete call set-up time in the MGC node	1810

V. ANALYSIS OF MEASURED RESULTS

Remote ISUP as an adapted ISDN User Part that is able to access the core network does not introduce significant delay into call set-up time. Nevertheless, interaction with external MG node that is necessary to transit from the collocated MG-MGC to splitted MG-MGC architecture causes additional delay. This additional delay was measured in the simulated environment and does not represent actual situation.

In case of Local-Local ISUP call MG-MGC is collocated on the both sides and there is no additional delay. The complete basic call set-up time in this node is 20 ms which is mostly spent on through connection in GS. The seizure of accesses on both sides in this case is almost momentary.

In cases of Local-Remote and Remote-Local ISUP call, when additional delay is neglected, complete measured basic call set-up time is 22 ms. Delay of 2 ms in comparison with previous Local-Local case is due to seizure of access for Remote ISUP.

Remote-Remote ISUP call through connection of bearers is performed in external switch, MG and complete basic call set-up time without simulations is 2 ms.

It is obvious that basic call set-up time is increased in the access node to the core network, at least for additional time used for interaction with dislocated MG. However, even when this additional time is neglected it can be noticed in the measurements that call set-up time has slightly increased.

VI. CONCLUSION

The ultimate goal of telecommunication networks is to have one multi-service network. However, the migration towards it should be smooth, cheap and compatible with existing networks. That implies separating of signalling and bearer and is the step forward towards one multi-service network. Introduction of this new splitted packet switched architecture had to be able to connect to existing networks in the cheapest acceptable way.

This connection in case of external circuit-switched ISDN/PSTN network is established using Remote ISUP functionality as shown in this case study. Its main advantage is the possibility to make interface to external (ISDN/PSTN) networks directly through core network Media Gateway (MG) instead of having MG inside AXE 10. Therefore, it reduces the amount of hardware needed what means lower expenses.

This article is concerned with the influence on the basic call set-up time by integration of packet switched and circuits switched core network using Remote ISUP. The results obtained by measurement shows when the additional delay due to interaction with dislocated MG is neglected, introduction of Remote ISUP functionality in the access node causes only small increase in basic call set-up time. Although this time is probably much longer than small increase obtained in this article, it is still small enough that reduced expenses of the network pay off this delay of basic call set-up.

REFERENCES

[1] Ericsson internal documents.

- [2] L.Bos, S.Leroy, "Toward an All-IP-Based UMTS System Architecture", *IEEE Network*, no. 1, p. 36, January/February 2001.
- [3] L. Becchetti, F.D. Priscoli, T. Inzerilli, P. Mahonen, L. Munoz, "Enhancing IP Service Provision over Heterogeneous Wireless Networks: A Path toward 4G", *IEEE Communications Magazine*, vol. 39, p. 74, August 2001.
- [4] K. Asatani, F. Bigi, P.A. Probst, "Telecommunications Standardization for The New Millenium: ITU-T's Strategies", *IEEE Communications Magazine*, vol. 39, p. 124, April 2001.
- [5] M. O. van Deventer, I.Keesmaat, P.Veenstra, "The ITU-T BICC Protocol: The Vital Step Toward an Integrated Voice-Data Multiservice Platform", vol. 39, p.140, May 2001.

ISBN 953-6042-87-8