Analysis of Call scenario in NGN network

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Abstract – Next Generation Network (NGN) is a safe technology for future. Market demands always change. Subscribers require different services with high quality. NGN network technology supports those services. In its efforts to keep the leading position in Kosova’s competitive telecommunication market, TK (Telecom of Kosova) is implementing its technological development strategy and migration to full IP NGN network. Scope of this paper is to describe the network connectivity in TK NGN. At the beginning is described the architecture of TK (Telecom of Kosova) NGN network with its corresponding elements and functionality. Such standard-based, open architecture core network enables the integration of Voice and Multimedia services on a single network infrastructure. Later, are described two call scenarios in NGN network in telecom of Kosova: NGN to NGN and NGN to PSTN (Public Switched Telephone Network) call scenario.

Key words: Next Generation Network, Voice over IP, Call scenario.

1 Introduction
NGN is a network that is designed for offering multimedia communications [4], which implies that it has broadband capacities, multichannel transport with high data rates, low latencies, low packet loss and QoS guarantees [1]. NGN represents new technology and services that all operators want to have at their disposal. NGN technology has been implemented in Telecom of Kosova, thus enabling converged IP/MPLS network services, focusing on opportunities for service differentiation and service-oriented technology. The convergent services are based on packet switching rather than circuit switching technology. In this paper we will describe elements connected to the NGN network in TK (telecom of Kosova), their functions and the requirements in terms of connectivity. Also, in this paper we will describe in details the scenario of SIP call, where 2 users are in NGN network, and the case where one user is in NGN and calls the user that is in PSTN network.

2 Elements of NGN network in TK and their connectivity
NGN in TK is composed of a number of elements; the most important elements are:

- Media Gateway
- Access Gateway (AGW) -1540 Litespan
- Trunk Gateway
- Signalling Gateway
- Softswitch-Alcatel 5020
- Media Server-CMS 1000
- Call Monitoring Node
- Service Router-SR
- Ethernet Service Switch-ESS
- BRAS (Broadband Remote Access Server)

In figure 1 is shown NGN [2] [3] network architecture in TK that illustrates the components of this network and their position within the network. Figure 1 also shows 7 LER nodes that are located in 7 main regions of Kosova. The Network Management System is connected through central LER located in capital city of Prishtina (upper part of figure) and this fact makes it more important than other LER (Label Edge Routers) nodes. Customers are connected to LER through AGW located in different zones, whereas LSR_A and LSR_B core nodes are connected with softswitch including firewalls in order to achieve a high level of system security from unauthorized interferences.
AGW nodes will be connected to the 7450 ESS switches through a 100Mbps Fast Ethernet connection. PSTN is connected with Softswitch through Signaling Gateway and to LSRs through Media Gateway. The traffic to and from the Internet passes through BRAS. NGN network topology in TK is based on:

1. VoIP Control Subsystem
2. Core/Edge Network
3. Access Network

### 2.1 VoIP Control Subsystems

Voip control subsystem consists of: Alcatel Softswitch platform 5020 - fulfills users’ requirements for voice and broadband multimedia services. Softswitch platform consists of four essential functions:

- Offers call control services and supports SIP, MEGACO/H.248 protocols (Call and Session Controller-CSC)
- Supports advanced interconnections with PSTN network (Media Gateway Controller-MGC)
- Keeps note in detail for user location (Subscription Location Server-SLS)
- Controls AGW-Litespans (Residential Gateway Controller-RGC)

### 2.2 Core/Edge Network

The core node of the NGN network of TK is based on feature rich Service Routers (LSR) and Ethernet Service Switch (LER) supporting all standardized protocols for IP/MPLS/Ethernet networks, guaranteeing QoS and respecting data traffic security.

- LER (Label Edge Router) – is technology that provides traffic services and enables Ethernet aggregation through IP/MPLS based network.
- LSR (Label Switch Router) – is a service router and flexible platform designed and optimized for data transport, voice and video services with high performances
- SAM (Service Aware Management) – is a network manager for 7xxx platform

### 2.2 Access Network

- Alcatel 1540 Litespan: is a multiservice access gateway that enables an operator to deliver ATM-based xDSL, TDM-based narrowband/wideband and NGN services to an area from a single node. The 1540 Litespan is a new generation DLC working as distributed Access Gateway.
- 1354 LMS (Litespan Management System) – managing element for voice and data services
in leased lines offered by 1540 Litespan platform, it manages narrowband services of Litespan.
- 5523 AWS (ADSL Work Station) – managing element for broadband services xDSL offered by LS 1540 platform, it manages broadband services of Litespan.

3. Types of services that are carried over NGN network:
In the Internet, data packets that belong to traffic flow are not guaranteed. Said in other words, the Internet is not suitable for real time applications, because network capacity is not guaranteed and delays are not limited. Thus, it is necessary to present an architecture that supports new services in the Internet and guarantees QoS for real time applications. NGN network in TK is able to support those services. Most important types of services that are carried over NGN in TK are:

OAM Traffic
OAM traffic is again subdivided in categories depending on the type of equipment:
- NGN OAM: OAM for/from NGN specific servers: A5020, A8690, A1300, A7510,
- Litespan OAM: to/from Litespans and their network management centers
- Network OAM: to/from routers, switches, firewalls and their network management

VOIP Traffic
VOIP traffic is a unique form of traffic in that it requires little bandwidth but very low latencies [5]. Call controlling protocols for VoIP traffic are: SIP, Megaco signaling and RTP voice traffic.
SIP & Megaco are signaling protocols that will be used to establish calls. Traffic will flow between terminal endpoints (AGW-Litespan, CMS1000 & A7510) and the NGN core equipments A5020 & A8690.
RTP traffic carrying the call will flow directly between the interested terminal endpoints (Litespan, A7510 & CMS1000).

Data Traffic
This traffic is pure Internet traffic between BRAS and DSL accesses in the Litespan & ISAM. This traffic must not be able to interact with the VoIP world but should be carried across the IP backbone from the ADSL (in Litespan or ISAM) directly to the BRAS. The Internet security and standard practices related to ADSL based access must be taken into account.

Supplementary services
Supplementary services that offers NGN network are: Call Forwarding services, Fixed announcements, presentation Services, Call screening, outgoing call barring, 3 party conference, Explicit call transfer, Call hold, Call waiting, PBX Line hunting

Intelligent services
Intelligent services in NGN network in TK are: Prepaid Card service, Freephone, Premium Rate, Televoting.

NGN SIP traffic
SIP technology is the key for NGN deployment. SIP is the IP based signaling used for the call handling of VoIP calls. Some of the equipment also uses the MEGACO signaling (gateways) but this other IP based signaling protocol will travel on roughly the same paths and within the same LANs as the SIP signaling. As it is a signaling protocol, SIP will flow between a terminal endpoint and the NGN core elements. Terminal endpoints are the Trunking Gateway, the announcement server and all Litespan nodes. NGN core elements are the A5020 VOIP, A5020 MGC and A8690 OSP.

NGN RTP traffic
RTP is the IP based protocol used to carry voice in NGN/VoIP networks. RTP traffic is sustained, composed of short UDP packets, heavy volume, requires high priority handling and small inter-packet delays in order not to impact the voice quality. RTP traffic always flows between 2 endpoint terminals; it never goes to the NGN core elements as these only handle signaling (SIP/MEGACO) flows.

4. NGN to NGN call scenario in TK
In order to see the call setup scenario in NGN network, it is presented one concrete system with all corresponding components that enable this process.
This process includes 5 steps as shown below (figure 2):
1. User A goes off-hook and dials the destination number. In this case a MEGACO message is sent from Litespan to the RGC that contains an off-hook event and dialed number. RGC sends a SIP message to the CSC server; this SIP message contains information on the originating point and number of the destination user (E.164). CSC proxies the INVITE message toward MMAS, where MMAS checks and applies originating side supplementary services if required (e.g. OCB-Outgoing Call Barring)
2. MMAS proceeds by sending an INVITE message toward CSC of the originating side (CSC_A). CSC forwards message to SLS to identify location of destination user. SLS looks up in its database and finds the CSC that hosts the user and sends status message 302 Temporary Moved (contains the ID of the CSC) showing the location of USER B (in our case CSC_A). CSC forwards INVITE message to MMAS to process supplementary services on the terminating user.

3. MMAS sends INVITE message to CSC. CSC_A figures it owns of the user and forwards the INVITE towards RGC_A. RGC_A finds user in its local database and then sends Megaco messages to the litespan of the destination user to set-up RTP context and applies ringing tones. In this case phone user B starts to ring. RGC_A sends status message 180 RING back to the CSC_A. CSC_A sends the status message back to the MMAS who originated the Invite. Now we have intensive communications between CSC and MMAS generating invite messages for originating side. Thereafter, CSC proxies’ status message to RGC who originated the Invite. RGC informs Litespan to complete the RTP path. The RTP path is then completed over the IP network from LS user A to LS user B. Now user A chats with User B.

4. Use B answers, RGC_A receives an off-hook event from Litespan. RGC_A sends status message 200 OK toward CSC_A. CSC_A proxies the reply back to MMAS (terminating side), MMAS terminating side, forwards it toward origin. After reciprocal communication between CSC_A and MMAS and generation of Invite messages, CSC proxies status message to RGC who originated the Invite. RGC informs Litespan to complete the RTP path. The RTP path is terminated.

5. After User B hangs-up, on-hook event is sent to RGC. RGC_A tells LS to remove RTP context for user B. RGC_A sends BYE message toward CSC_A to terminate the call. Now we have a intensive communications between CSC_A and MMAS through call set-up but in opposite direction. Then the CSC proxies BYE message to RGC who originated the Invite. RGC tells LS to remove RTP context and apply tone. Call is terminated.

5. RGC to PSTN call Scenario in TK

Call setup scenario between NGN and PSTN network includes 4 steps as below (figure 3):
1. User A goes off-hook and dials the number of user B. Invite message is sent with originating user SDP towards CSC (SDP contains RTP information for user A). CSC Proxies message towards MMAS, where MMAS applies originating side supplementary services. MMAS initiates terminating call leg and sends Invite message toward CSC. CSC_A receives Invite and determines it doesn’t know the terminating user. CSC_A forwards message towards SLS for subscription location identification. SLS looks up in its database and finds that it doesn’t know the user. In this case SLS tells CSC_A that the user A is unknown. CSC_A checks next destination in routing table: forwards message to MGC towards PSTN network. MGC translates request into N7 IAM message and forwards it to the PSTN network.

2. MGC receives ACM message, indicating user B has been reached (and is ringing). MGC translates message into 180 ringing status message towards MMAS, which will in turn forward to the originating RGC. RGC_A informs LS to remove tone and set-up RTP context.

3. RTP path is set-up and user A talks to user B.

4. User B hangs up, PSTN network sends REL message which is the translated from MGC into BYE message and is send to MMAS for processing. MMAS in turn proxies the BYE message towards the originating call leg. RGC_A tells the Litespan to break down RTP flow and set tone.

6. Summary

With growing demands for new services, Telecomm of Kosova is forced to adopt open technologies for offering different services. This enables a fast response to market demands. In this paper is described in detail the NGN network architecture. Also are described in detail two call scenarios: one call scenario inside NGN network, another one is between NGN network and PSTN network.
7. References


