NEXT GENERATION NETWORKS ARCHITECTURE FOR MULTIMEDIA APPLICATIONS

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Abstract: Demand for new services and application is the main driver for entire network and terminal evolution. New forms of communication emerge from combining different media in applications used by end-systems and users. A new concept of horizontally layered architecture is being standardized through 3GPP and other bodies. Separated application, control and connectivity layers allow independent evolution as market and technology develops. Session initiation protocol (SIP) is the standard control protocol for establishing multiparty, multimedia communication in IP-based networks. The architecture of next generation multimedia network based on 3GPP standardization of IP Multimedia Subsystem (IMS) is described. At the end of the article new end-user services are illustrated.

Key words: SIP, multimedia services, IP multimedia subsystem, layered network architecture

1. INTRODUCTION

As traditional voice service is becoming insufficient for users’ requests for new, diverse and multimedia applications, a new approach in building communication infrastructure is required. This new infrastructure approach called New Generation Network (NGN) complies with several general requirements put upon future communication networks. It provides support for new ways of communicating created by advanced user terminals and Internet related technologies, which can be video conferencing, collaborative working, streaming multimedia, presence, messaging etc. This new Multimedia Communication will offer new properties in people communication, like richer communication experience, more senses engagement and "showing" rather than "explaining".

Another important feature of such architecture is its ability to provide new multimedia services to users everywhere regardless of access technology, fixed or mobile. Thus, it is prepared for services and applications that can be developed for today or next generation networks.

NGN Multimedia architecture is based on the 3GPP, 3GPP2 and IETF standardization work and uses the Session Initiation Protocol (SIP) [1] for control signaling as it’s features positioned SIP to be the preferred technology to address NGN communications. NGN Multimedia solution is based on horizontal layered architecture (Fig. 1) that allows independent development of each layer as market and technology develops. One of the most significant advantages of this independent development approach is easy migration of every layer to the new technologies without need for changing other layers.

The applications layer consists of application and content servers for executing value added services to the end user. The control layer manages session setup, modification and
release, and can also perform mobility management, security, charging and interworking towards external networks. Routers, switches, media gateways and other user plane elements make the connectivity layer.

**Application Layer**

- Service and Content Entities

**Control Layer**

- Session Control Entities
- Gateway Entities

**Connectivity Layer**

- IP Network
- Broadband Access

**Terminals and Clients**

*Fig. 1. NGN Multimedia layered architecture*

In the following chapters, NGN Multimedia architecture with its components and services it provides will be described in more detail.

### 2. SIP

The Session Initiation Protocol (SIP) is an Internet standard specified by the Internet Engineering Task Force (IETF) that is used to initiate, manage, and terminate interactive sessions between one or more participants. It is simple and open protocol supporting interworking with ISUP and H.323 that enables integration of PSTN with IP networks and also gives service providers the ability to offer new services that can go well beyond Voice over IP (VoIP).

SIP is an application-level control protocol, which defines syntax and semantics for the messages to be exchanged between network elements and controls end-to-end session establishment. Figure 2 depicts SIP's relationship to other protocols used in the Internet.

*Fig. 2. Relationship of various protocols used on the Internet*

SIP is a text-based protocol which reuses message structure found in the hypertext transfer protocol (HTTP) and simple mail transfer protocol (SMTP) with numerous informational headers followed by a body and uses Session Description Protocol (SDP) for media description. SIP also defines different entities, namely the SIP User Agents (UA), SIP server and the Registrar. UAs could be separated in UA Servers and UA clients. UA clients initiate a call, while UA servers only react to calls. SIP server or Call/Session Control Function (CSCF) is the main element in SIP architecture and offers a platform for
implementing next-generation communication services in multimedia-enabled networks [1], [2]. SIP Registrar stores the registration information received from User Agent in a location service. When the information is stored, the Registrar sends the appropriate response back to the user agent.

Some examples of services supported by SIP architecture are adding or removing new media types in the middle of a session, simultaneous user registering in multiple locations, presence and instant messaging, conferencing and distance working, VPNs and others [2].

3. SOLUTION ARCHITECTURE

The network architecture of IP Multimedia System is based on 3GPP standards [3]. The functionality is separated into three horizontal layers – an application layer, a control layer and a connectivity layer. Figure 3 shows logical entities and communication protocols, which are building blocks of the NGN multimedia architecture. Many logical entities can be implemented and executed on the same hardware platform.

![Figure 3. NGN multimedia layered architecture – entities and protocols](image)

The control entities, which are central in session establishment, are accessed by terminals and clients via the connectivity layer. Terminals and clients use applications implemented in service and content entities residing in the application layer. Gateway entities provide interworking with circuit switched telephony networks and the public Internet. Connectivity layer consists of broadband access networks and a backbone network acting as a bearer.

3.1 Terminals and clients

End users can access services via terminals such as PCs, personal digital assistants (PDA) or new SIP-based telephone devices. Clients can also exist in set-top boxes and residential gateways creating home networks. Terminals and clients are expected to support
basic functionality such as: IP and security functionality as required by IP bearer network, SIP and Session Description Protocol (SDP) as defined by IETF and 3GPP, and SIP based presence and instant messaging applications.

3.2 IP bearer network

The IP bearer network is responsible for carrying the end user traffic between endpoints of the network. It also connects terminals and clients to the session control and application entities. IP bearer network can be divided into an access network, using ADSL, Ethernet or Wireless LAN as transmission technologies, and backbone part with associated DNS, DHCP and AAA servers in the application layer. Quality of service (QoS) is a complex issue that must be considered in all entities along the connection path, including the capabilities of the terminals at both ends.

3.3 Call Session Control Entities

Control layer consist of call/session control servers. It is in charge of set-up, modification and release of calls and sessions by manipulating the resources in the connectivity layer. It also handles functions like mobility, security and charging, and allows triggering of services in the application layer.

Call Session Control Function (CSCF) servers are core nodes of NGN Multimedia architecture. They are also referred to as a SIP server. CSCF supports establishment, modification and release of IP multimedia sessions using the SIP/SDP protocol suite. CSCF can be configured for three different roles in the network: Proxy Call Session Control Function (P-CSCF), Interrogating Call Session Control Function (I-CSCF) and Serving Call Session Control Function (S-CSFC) [3].

P-CSCF is the initial point of contact for the terminal in the control layer. It forwards SIP messages received from terminal to a SIP server in the home network. P-CSCF can modify outgoing request according to a set of rules defined by the network operator and it can also be used as a default inbound and outbound proxy for address translation services, guaranteeing QoS and invocation of locally-significant services such as taxi locators, weather or news.

I-CSFC resides in the user's home network and is a contact point for all connections destined to a subscriber or a roaming subscriber currently inside operator's service area. It hides inner topology of the home network and determines and locates appropriate S-CSFC.

S-CSFC supports establishment, modification and release of IP multimedia sessions. It provides service invocation, either directly through some local execution function or indirectly through an API towards the Services/Applications layer. Capability negotiation at session invocation is performed at S-CSFC. Functionality of S-CSFC includes routing and redirection of originating sessions to external networks and routing of terminating sessions to visited networks, modification and clearing of multimedia sessions and subscriber registration.

Home Subscriber Server (HSS) [3] is a master database that stores user and subscriber profiles and keeps track of core network nodes that are handling the subscriber. User data are downloaded to S-CSFC, while HSS stores temporary data with the location of the S-CSFC where a user is registered. It supports user authentication and authorizes user service requests.

DNS/ENUM entity translates the different public identifiers of the user. It provides name to address translation required by the call session control entities.
3.4 Media resource function (MRF)

MRF [3] supports multi-party multimedia conversations, message playing and media conversion. It is decomposed into control, MRF Control part (MRFC) and connectivity, MRF Processing part (MRFP). MRFC controls and selects the MRFP using H.248 or Real-Time Streaming Protocol (RTSP). CSCF and application server use SIP and vXML respectively to invoke and instruct MRFC. MRFC executes vXML scripts to provide conferencing services to the caller. It has the ability to create ad hoc conference sessions, schedule conference session, add and delete conference members. Set of codecs, transcoders and mixing functions are implemented in MRFP to enable audio and video manipulations in the connectivity layer. MRFP is instructed by MRFC to connect media resources to media streams. The MRFP is divided into media bridges for real-time applications (conference bridging and transcoding) and media player for streaming applications (voice mail, network announcements and information services).

3.5 PSTN Gateway entities

PSTN Gateway is decomposed into Media Gateway Control Function (MGCF), Media Gateway (MGW) and Signaling Gateway (SGW) [3]. It enables interworking of circuit-switched telephony network and multimedia domain. MGCF controls its slave gateways SGW and MGW. It terminates call control signaling (ISUP) from SGW and instructs MGW how to provide inter-working in the connectivity layer. MGCF handles multimedia session establishment, modification and termination using SIP protocol in multimedia domain and e.g. ISUP or Q.931 in the PSTN domain. It also supports addressing and routing of multimedia sessions to and from CSCFs and PSTN nodes. MGW adapts payload from PSTN circuits into IP packets suitable for transport over IP bearer network. SGW provides bridge between SS7 network and IP network. It encapsulates ISUP messages using the thin-ISUP protocol over a TCP connection. ISUP messages are then transported to MGCF that maps them into SIP messages for signaling in multimedia domain.

3.6 Service network and associated entities

Service network consists of open, IP-based applications and standardized protocols. Closely related to the service network is the Service Creation Environment (SCE). SCE provides a framework for development of multimedia applications and includes set of resources, tools and documents supporting developers. Application server is located in the applications layer and is based on platform independent Java technology. Presence server supports presence-based services using IETF’s SIMPLE methodology. Provisioning portal allows end-users to manage the behavior of their services through HTTP interface. The end-user can modify provisioning data through this interface directly.

4. END-USER SERVICES

The services offered by NGN networks can broadly be grouped into elementary services and combinations of elementary services. The elementary services inherent in the described NGN architecture are:

- Conversational Multimedia Sessions, two party and multi-party
- Presence Information Handling. The presence service keeps track of the end user states in the network, if the user is registered. A presence service allows users to subscribe to each other and be notified of changes in state. A Presence service could be combined with other services, for example checking the state before trying to set-up the session.
- Streaming Multimedia, live, i.e. broadcast, and on demand
- Messaging
- Session Control Capabilities

Using these elementary services NGN architecture offers the end-user a range of services as listed in table I. Additional end-user services can also easily be created by the operator or by 3rd party developers.

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<th>Table I NGN, combined services</th>
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<tr>
<td>Conversational Services</td>
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<td>Audio and video two party sessions</td>
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<tr>
<td>Multi-party conferencing</td>
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<td>Personalized Inbound Session Handling</td>
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<td>Unified Messaging</td>
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The majority of the services provided to subscribers in the multimedia domain require that the end-user is connected to the network using a SIP User Agent (UA) terminal.

5. CONCLUSION

The building blocks for multimedia services have been standardized in 3GPP. Integration of IP technology into the core network is dictated by the need for a common transport technology and the support for new services. Quality of service, resilience and security mechanisms in IP should be enhanced to provide carrier-class characteristics of the connectivity network. Changes in the way people communicate will be introduced by enhanced services with rich media. Success of these improvements will depend on implementation of standardized architectures and quality of multimedia content.

REFERENCES