Requirements and Service Scenarios for QoS enabled Mobile VoIP Service

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Abstract. The QoS enabled mobile VoIP service has the characteristics of ensuring the QoS when the user terminal moves from one access point to another one. This QoS enabled service requires some form of adaptation in order to support service continuity when users' requirements and network conditions mismatch by negotiation of network QoS and/or terminal parameters as codec change or adaptation. The requirements in user terminal, network and media handling aspects are described, along with service scenarios when user terminal moves to different access networks between different operator domains, and these proposed service scenarios will became useful guidelines for implementation of QoS enabled mobile VoIP service in mobile operator's network.

Keywords. VoIP, Mobility, WiFi, 3G, 4G-LTE

1. Introduction

The mobile Voice over IP service is assumed to be a seamless service, and the QoS enabled mobile VoIP service is a mobile VoIP service which has the characteristic of ensuring the QoS when the user terminal moves from one access point to another one. This service requires some form of adaptation in order to support service continuity when users' requirements and network conditions mismatch. Adaptation may include negotiation/renegotiation of network QoS and/or terminal parameters as codec change or adaptation. When the user terminal moves between different networks, network QoS parameters may either be maintained or adapted. Network QoS parameters include information such as IP layer available bandwidth, maximum packet transfer delay, jitter, packet loss rate, burst loss rate, packet delay variation tolerance and packet per second, etc.

Figure 1 illustrates a general network architecture involving two operators supporting different types of access networks such as 3G, WiFi, and 4G-LTE and where users of the mobile VoIP service may move between different access networks in the same operator domain or between different operator domains.

2. Requirements in user terminal, network and media handling aspects

Some definitions and necessary requirements for providing the QoS enabled mobile VoIP service in user terminal aspects, network aspects, media handling aspects, and end-to-end aspects are as follows.

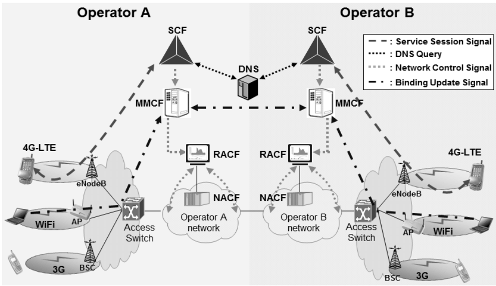

Fig. 1. General network architecture for QoS enabled mobile VoIP

2.1 User terminal aspects

The user terminal (UT) is the device that facilitates end user's access to services and applications. User terminals can be classified as single-mode terminals and multi-mode terminals.

- Single mode terminal: a terminal which provides connection to either fixed or mobile or a wireless network. Connection of a single-mode terminal to the network is identified by a single identity which is the basis to bind all services and applications delivered to the terminal. A single-mode terminal is equipped with various media processing capabilities including various codecs to handle services and applications, such as voice phone call, conference, SMS, video and IPTV.
- Multi mode terminal: a terminal which provides connections to fixed, mobile or wireless networks. Each network connection of a multi-mode terminal is identified by a different identity according to each network. Therefore binding of services and applications delivered to a multi-mode terminal is much more complex than for a single-mode terminal. A multi-mode terminal is also equipped with various media processing capabilities including various codecs to handle services and applications similar to those for single-mode, but forming different set of service categories with different set of processing capabilities such as different set of codecs according to associated network connection capability. User terminals considered in this paper are multi-mode terminals, equipped with WiFi, 3G, and 4G-LTE functions and different voice codecs can be associated.

2.2 Network aspects

The networks considered in this paper are WiFi, 3G, and 4G-LTE networks, and a user terminal connected to different access networks may use different service coverage and different transmission speed. These networks have different capabilities owing to the different available bandwidth, service coverage, and transmission speed when a user terminal moves or accesses these different networks. The following identifies the main characteristics of these networks:

- WiFi: this network provides wireless access technology approximately 10 Mbps ~ 100 Mbps (802.11x based) transmission speed to user terminals as personal digital assistant (PDA) or notebook within several hundred meters from user terminal to access point.
- 4G-LTE: this network provides wireless access technology approximately 50 Mbps ~ 100 Mbps transmission speed to user terminals as smart phone or smart PAD or Tablet PC within 5km ~ 100km service coverage per cell.
- 3G: this network provides wireless access technology approximately 10 Mbps ~ 50 Mbps transmission speed to user terminals as PDA or notebook with 250 km/h mobility support within 1km ~ 3km service coverage per cell.

This paper assumes the use of Layer 2 (e.g. MAC) authentication based on triggering of Layer 2 signal strength in the user terminal.

2.3 Media handling aspects

The following two aspects can be considered when a user terminal moves between different networks: same quality and different quality

- Same content quality: content quality remains the same whenever the user terminal moves to between different networks. In this case, there is no need to change of media processing capability and network QoS parameters remain the same.
- Different contents quality: This is the case where content quality may change when the user terminal moves between different networks. In this case, there may be a need to change the media processing capabilities as well as network QoS parameters.

This paper assumes that content quality can be changed when a user terminal moves between different networks and as a result, different network QoS parameters may be applied in these different networks.

2.3 End-to-end considerations

Figure 2 shows the overall considerations made in this paper regarding user terminal, networks, and content, and illustrates the following considerations:

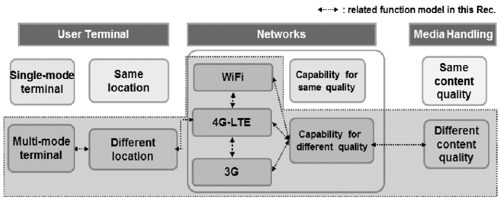

Fig. 2. End-to-end view of key assumptions

- a multi-mode user terminal is operating such that its network connection is kept active as long as possible;
- the network connection of a user terminal may change depending on the user terminal behavior such as when moving or changing the connection among WiFi, 4G-LTE, and 3G networks. Changing of access network implies a change in the connecting capabilities like bandwidth, security, QoS, service coverage and others. Therefore, networks can be characterized by two elements: connection change and capability for different quality.
- content quality can stay the same or can change (i.e. downgrade or upgrade) depending of end-to-end QoS conditions involving the user terminal and networks. For example, when a user terminal moves from a WiFi network to 4G-LTE network during VoIP service, the QoS enabled VoIP service may continue by QoS profile transfer from WiFi network to 4G-LTE network.

3. Service scenario for QoS enabled mobile VoIP service

The service scenarios by considering the case where the user terminal moves:

- in a single operator's network which supports either WiFi or 3G or 4G-LTE;
- in a multiple operator's networks, each supporting a different wireless network, e.g. WiFi, 4G-LTE or 3G wireless network;
- in a single operator's network where the user terminal moves between different wireless access networks, e.g. between a 4G-LTE and a WiFi network.

Here, only second case are described for service interworking between different operator's networks such as between a WiFi network and a 3G network, and these scenarios include registration, service establishment, data transfer configuration, and service release.

3.1 Registration

The attachment to the network (IP address allocation) and service provider registration (e.g. IMS network registration) is performed, so registration includes:

- network attachment procedures including temporary IP address allocation and registration of the binding between the persistent IP address and the temporary IP address;
- service stratum authentication and authorization procedures based on information included in the service user profile.

Figure 3 illustrates the overall procedure related to the registration.

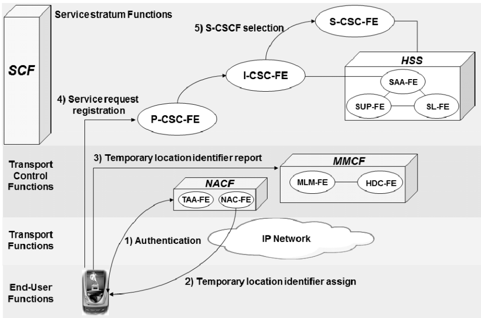

Fig. 3. Registration

The following describes in more detail the information flows shown in Figure 3:

1) L2 MAC layer authentication and authentication, authorization, accounting (AAA) associations are executed by the TAA-FE after connecting to the network;
2) A temporary IP address (network IP address) is assigned to the user terminal by the NAC-FE of NACF;
3) The received temporary IP address from local network is reported by the user terminal to the MLM-FE;
4) A service request registration message which contains the user's URL is sent to the P-CSC-FE by the user terminal;
5) The S-CSC-FE sends a service request registration message which contains the user's URL and P-CSC-FE URL to the SL-FE via the I-CSC-FE.

3.2 Service establishment

Service establishment covers the required signaling information for call establishment as shown in Figure 4, and described in more detail as follows: .

1) The originating user terminal forwards a service request message the P-CSC-FE of operator A. This message contains the URL information of originating and terminating user terminals, codec information and UDP port numbers;
2) The I-CSC-FE of the SCF of operator A interprets the service request message received from the originating user terminal;

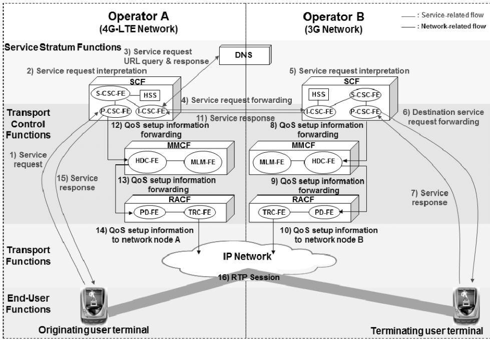

Fig. 4. Service establishment

3) The I-CSC-FE of SCF of operator A forwards the URL (Uniform Resource Locator) of the service request message to the DNS (Domain Name System) and gets in the response the network IP address information of the terminating user terminal after recognizing that the URL (Uniform Resource Locator) of the terminating user terminal does not correspond to its own domain name;
4) The I-CSC-FE of operator A forwards the service request message to the I-CSC-FE of operator B after getting the SIP URL of operator B;
5) The I-CSC-FE of operator B interprets the service request message which was received from the I-CSC-FE of operator A;
6) The service request message from the originating user terminal is forwarded to the terminating user terminal;
7) The terminating user terminal forwards the service response message the P-CSC-FE of the SCF of operator B. This message contains the URL information of originating and terminating user terminal, codec information and UDP port number of terminating user terminal;
8) In the operator B network, the P-CSC-FE forwards the QoS set up information (which contains network bandwidth information) to the PD-FE;
9) The TRC-FE of operator B requests QoS set up information to the IP network elements of operator B;
10) The I-CSC-FE of operator B forwards the service response message to the I-CSC-FE of operator A;
11) The P-CSC-FE of operator A forwards QoS set up information to the PD-FE of operator A;
12) The TRC-FE of the RACF of operator A requests QoS set up information to the IP network elements of Operator A;
13) The originating user terminal receives the service response message from the P-CSC-FE of operator A.

3.3 Data transfer configuration

Data transfer configuration includes the use of the real time transport protocol (RTP) for support of voice service as shown in Figure 5, and described in more detail as follows:

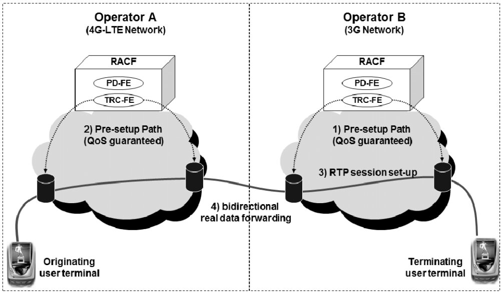

Fig. 5. Data transfer configuration

1) The RACF of operator B sets-up the pre-established path in order to maintain the same level QoS that the one requested during the initial call establishment;
2) The RACF of operator A sets-up the pre-established path in order to maintain the same level QoS that the one requested during the initial call establishment;
3) An RTP session is set-up between the originating and terminating user terminals;
4) Bidirectional real data is forwarded between the originating and terminating user terminals.

3.4 Service release

Service release includes the information required for call termination as Figure 6, and described in more detail as follows:

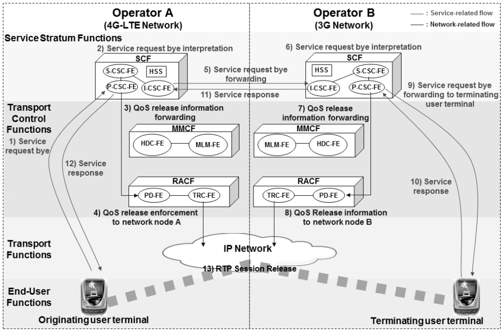

Fig. 6. Service release

1) The originating user terminal sends a BYE service request message to the P-CSC-FE of operator A;
2) The I-CSC-FE of operator A interprets the BYE service request message which was received from the originating user terminal;
3) The P-CSC-FE of operator A forwards the QoS release information to the PD-FE of operator A;
4) The TRC-FE of operator A forwards the QoS release message to the IP network elements of operator A;
5) The I-CSC-FE of operator A forwards the BYE service request message to the I-CSC-FE of operator B;
6) The I-CSC-FE of operator B interprets the BYE service request message which was received from the I-CSC-FE of operator A;
7) The P-CSC-FE of operator B forwards the QoS release information to the PD-FE of operator B;
8) The TRC-FE of operator B forwards the QoS release message to the IP network elements of operator B;
9) The P-CSC-FE of operator B forwards the BYE service request message to the terminating user terminal;
10) The terminating user terminal sends a service response message to the P-CSC-FE of operator B;
11) The I-CSC-FE of operator B forwards the service response message to the I-CSC-FE of operator A;
12) The P-CSC-FE of operator A forwards the service response message to the originating user terminal;
13) The RTP (Real Time Transport Protocol) session is closed.

4. Conclusions

Requirements in user terminal aspects, network and media handling aspects are mentioned, and service scenarios are described in registration, service establishment, data transfer configuration and service release to provide QoS enabled mobile VoIP service by supporting seamless mobility among WiFi, 3G, 4G-LTE networks. Nowadays, the VoIP service has been launched in operator's WiFi, 3G, 4G-LTE networks, and seamless mobility among these networks are requisite when user terminal moves to the different networks, so proposed requirements and service scenarios will became useful guidelines for QoS enabled mobile VoIP service in near future.

References

1. ITU-T Recommendation Y.2001, General overview of NGN
2. ITU-T Recommendation Y.2201, NGN release 1 requirements
3. ITU-T Recommendation Y.2012, Functional Requirements and Architecture of the NGN
4. ITU-T Recommendation Y.2021, IMS for Next Generation Network
5. ITU-T Recommendation Y.rtconv, IMS based Real-time Conversational Multimedia Service over NGN
6. ITU-T Recommendation Y.2211, IMS-based real-time conversational multimedia services over NGN.
7. IETF RFC 4886 , Network Mobility Support Goals and Requirements.
8. IETF RFC 5026, Mobile IPv6 Bootstrapping in Split Scenario.
9. IETF RFC 5149, Service Selection for Mobile IPv6.
10. 3GPP TS 33.234 V8.1.0, 3G security; Wireless Local Area Network (WLAN) interworking security.