Source: Samsung Electronics Co. Ltd

**Title: [FS\_eiRTCW] Aspects to consider during FS\_eiRTCW study**

**Agenda Item: 10.6**

**Document for: Discussion and Agreement**

# **Introduction**

During the SA4#118e meeting, couple of contributions [1] (covering the collaboration scenarios for WebRTC) and [2] (defining the scope of FS\_eiRTCW) were agreed as part of the FS\_eiRTCW study. The permanent document (PD) for FS\_eiRTCW defines the skeleton of topics that needed to be studied as part of this study. This contribution discusses some aspects related to architectural components and C-plane signalling protocol requirements that need to be considered while developing the architecture related to this study.

# **Aspects to consider in this study**

The study item proposal and the PD for FS\_eiRTCW clearly establish the motivations for native WebRTC signalling instead of the over-engineered SIP protocol. One of the objectives of this study is to look into architecture components and C-plane signalling requirements to develop a lightweight signalling protocol. While we look into these requirements, there are few aspects that need to be considered.

#### 2.1 WebRTC service operations

The PD and study item proposal references TS 24.371 in this study. TS 24.371 specifies WebRTC access to IMS core. While this TS does not describe the details of the C-plane signalling, it provides details about how WebRTC clients can access the IMS subsystem:

* Registration and authentication: Clause 6 of TS 24.371 describes aspects related to registration and authentication of WIC (WebRTC IMS client) in the IM CN subsystem that are required for support of WebRTC. In specific, the roles of WIC, WWSF, and eP-CSCF are specified for registration aspects such as WIC registration of individual Public User Identify using IMS authentication, WIC registration of individual public user identify using web credentials, and WIC registration of individual public user identify from a pool of public user identities.
* Deregistration: Clause 6A of TS 24371 specifies deregistration aspects and the role of WIC and eP-CSCF for this procedure
* Call origination and termination: Clause 7 of TS 24371 specifies the roles of WIC, WWSF, and eP-CSCF during call origination and termination procedures such as origination of call by the WIC, termination of call by WIC, and emergency calls by WIC
* Data channel open and close: Clause 8 of TS 24371 specifies the roles and responsibilities of WIC, WWSF and eP-CSCF in data channel operations
* Call modification: Clause 9 of TS 24371 specifies the roles and responsibilities of WIC and eP-CSCF for call modification procedures

TS 24371 specifies the above procedures for WIC client access to IMS core. These procedures are primarily based on SIP protocol. So, if the objective of the FS\_eiRTCW is to define a lightweight C-plane signalling protocol, this protocol has to support the above WebRTC service functionalities at the very minimum.

#### 2.2 Collaboration Scenarios

[1] described four collaboration scenarios for WebRTC with an operator network:

* Collaboration scenario 1: 5G support for OTT WebRTC
* Collaboration scenario 2: MNO provided trusted WebRTC function
* Collaboration scenarios 3: MNO facilitated WebRTC services
* Collaboration scenario 4: Interoperable WebRTC services

As described in [1], the first two collaboration scenarios 1 & 2 involve an OTT provider hosting the WebRTC signalling server at minimum. Collaboration scenarios 3 & 4 describe operator provisioned or managed WebRTC services. It is not clear what collaboration scenarios will be studied as part of the FS\_eiRTCW work. However, following the discussion during the SA4#118e meeting, and the work structuring of FS\_eiRTCW and iRTCW [3], it seems collaboration scenarios 1 & 2 are applicable to iRTCW normative work, and collaboration scenarios 3 & 4 apply to this FS\_eiRTCW study.

It is recommended that the appropriate collaboration scenarios for FS\_eiRTCW study are documented in the PD document so there is clarity during the architecture development stage.

#### 2.3 Security considerations for Interoperable WebRTC services

This aspect focuses on collaboration scenario 4 discussed in [1], where a globally interoperable WebRTC service is provided, and mobile users from different MNOs are able to join the same service and benefit from the 5G system support for better end-to-end quality of service. When an interoperable service is provided that can be accessed by users of different MNOs, one of the aspects that need close consideration is the aspect of authentication and authorization.

TS 24371 briefly discusses authentication and authorization aspects wherein the WWSF forwards the auth tokens to the WIC client that are issued by the WAF. For an interoperable WebRTC service, it needs to be studied whether each MNO individually authenticates the users to access the MNO service, or if the MNOs use a shared authentication service so the users only authenticate to that shared service (e.g., using Oauth) and the authentication and authorization information is shared with the MNOs.

In addition to authentication and authorization, key management aspects because of multiple MNOs for interoperable WebRTC services are to be looked into.

It is probable that such mechanisms are not studied as part of the FS\_eiRTCW study in 3GPP SA4, but instead left to referring to existing 3GPP SA3 specifications. If no existing mechanisms specified by 3GPP SA3 can be used for this aspect, then there may be a need to liaise with 3GPP SA3 for the development of such an architecture.

#### 2.4 WebRTC function deployment options

TS 24.371 describes two functions that help with WIC (WebRTC IMS client) access the IMS core – the WWSF and the WAF. [1] also presents network functions necessary for delivering WebRTC services – the WebRTC signalling server, and the complementing servers such as STUN, TURN, MCU etc. With an objective of delivering WebRTC services using the 5G architecture, an initial mapping of WebRTC functions to 5G functions can be done to help with architecture development.

In 5G media architecture, there are mainly two types of functions – an AF (application function) and an AS (application server). AF operates within the control plane of the service to help with service setup (provisioning, management etc.) and an AS helps with the user plane i.e. to deliver the service to end users. An initial mapping of WebRTC functions described in [1] can be made as below:

* WebRTC signalling server: The WebRTC signalling server helps with session setup of WebRTC sessions. [3] indicates that the WebRTC signalling server interfaces with PCF for QoS management. Two deployment options are possible:
  + A separate WebRTC server interfacing with a 5G AF. With this option, a separate interface between the WebRTC signalling server and AF has to be specified. This may apply to the collaboration scenarios 1 & 2 described in [1] and may be out of scope of FS\_eiRTCW study, but in scope for the iRTCW normative work.
  + WebRTC signalling server is implemented as a 5G AF so existing AF interfaces with AS, PCF, NEF still hold. This option is more desirable as the AF capabilities specified in TS 23501, TS 23502 can be used such as AF influenced traffic routing and QoS management by interfacing with PCF
* Other servers such as STUN, TURN, MCU etc: These functions do not have a direct role in session management, so these functions are likely deployed as AS functions

Having a clear mapping between 5G media delivery architecture components defined in TS 26501 may be helpful while working on the architecture development for FS\_eiRTCW

#### 2.5 Use 5GMS architecture enablers for WebRTC sessions

One of the main objectives of the study is to define a new lightweight C-plane signalling protocol instead of using SIP. For this objective, it may be useful if enablers from existing 5G media streaming architecture can be used such as the following:

* TS 26501 and TS 26512 specify network assistance enabler where the network can help the UE in throughput recommendation and delivery boost capabilities. Can network assistance be applied to WebRTC sessions thereby enhancing the service experience of WebRTC services?
* Traditional WebRTC QoS relied on best effort QoS and DSCP marking, either by the browser or a CPE. If WebRTC signalling server interfaces with PCF for QoS management as specified in [3], can dynamic policy and QoS management aspects defined using M1 provisioning API (in TS 26501 and TS 26512) be used for WebRTC services?
* One way of WebRTC flow identification for differentiated QoS is by relying on TURN server based WebRTC where the TURN server has mapping of RTP flows. Recently, some work was done for traffic identification for media flows and documented in clause 5.3 of TR 26804. Can such traffic identification help identify WebRTC related flows so QoS of those WebRTC flows can be managed better?
* 3GPP SA4 group is also studying network slicing extensions for media streaming services. Can network slicing help with better provisioning and QoS management of WebRTC traffic?

While performing study of FS\_eiRTCW, it may be useful to study the relevance of existing 5GMS architecture enablers to better support WebRTC services.

# **Potential open issues for FS\_eiRTCW study**

* 1. The C-plane signalling protocol should support basic WebRTC service operations such as client registration, authentication and authorization; call control; and data channel management that are relevant to the new architecture.
  2. Security considerations for interoperable WebRTC services such as authentication, authorization, and key management
  3. Deployment options of traditional WebRTC functions in 5G network, and mapping of those functions to 5G media architecture

NOTE: Mapping of WebRTC functions to 5GMS functions to be confirmed in 5GAREA study

* 1. Feasibility to use existing 5GMS architecture enablers for betterment of WebRTC services.

# **Proposal**

We propose that the five aspects described in clause 2 be considered during the study, and request adding clause 3 to the PD.

# **References**

[1] S4-220519, “Considerations on WebRTC QoS architecture”, SA4#118e, April 2022

[2] S4-220420, “Proposal for FS\_eiRTCW”, SA4#118e, April 2022

[3] S4-220517, “Structuring work on iRTCW and FS\_eiRTCW”, SA4#118e, April 2022