ALTICE LABS WHITEPAPER

Continuity of services and mobility

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Contents

Summary

The success of mobile networks presents a paradox: users generally accept lower voice quality and service at the expense of obtaining a service feature that offsets these losses: mobility! Thanks to complex control mechanisms, it is possible, within certain conditions, to ensure the continuity of service sessions while the user, possessing a certain access terminal, moves freely. In this scenario we can, in reality, identify terminal mobility through which the user can benefit from its mobility. This can be complemented with session and service mobility.

The use of the term mobility is very often used excessively, meaning rather nomadism, that is, the service is available in different physical locations and timelines, but unavailable while the transition is running. The continuity of service, usually associated with some sort of mobility, is a feature that ensures, in the context of telecommunications, the enriched transfer of sessions without disruptions that are perceptible to the user.

Given that users are becoming increasingly more accustomed to the use of an increasing number of different terminals, at home, on the street, and in the office, there is a natural need and the opportunity to use the most suitable device for each component of multimedia communication. Continuity of service provides a guarantee of uninterrupted multimedia sessions, together with the types of mobility referred, by for example adding the flexibility to transfer multimedia sessions from one to several other different devices, e.g. decomposing it into its audio and video components.

This document aims to identify the main solutions for guaranteeing session continuity, associated with the various types of mobility, with the emphasis on LTE/EPC networks.

1. Introduction

It is now undisputed that the evolution of the whole of the telecommunications networks is to be done in the sense of the widespread adoption of the IP as the base element of communications (All-IP). Access technologies, such as LTE and GPON, accelerate this process. In addition, at the level of services, the IMS is also a key element which should gradually support the majority of telecommunications services, as part of an overall development strategy of the current legacy networks, fixed and mobile, towards a NGN (Next Generation Network) architecture, providing universal services. The use of common, standardised services' platforms on IMS, irrespective of the access technology used, allows the desired FMC (Fixed-Mobile Convergence) to be achieved, giving the individual and the services a high level of mobility, depending on various aspects.

In this context, some aspects emerge as more evident features, highlighting, from the point of view of users, the availability and increasing bandwidth consumption and preference for the use of telecommunication services through the use of personal mobile terminals, supporting multiple network access technologies. Session continuity enables fixed-mobile convergence, as well as the use of multiple devices and multiple networks.

These are firm steps with a view to achieving the concept of ABC (Always Best Connected), in that the user uses telecommunications and data services, incorporating various mechanisms for mobility and continuity of its service sessions.

2. Framework technologies: LTE/EPC and IMS

LTE/EPC [TS 36.300] and the GPON, as well as the DSL, are access technologies, connecting the end-user to communications networks. They work in quite distinct domains, fibre access and radio access respectively, but have two common characteristics: the availability of high bandwidths and use of the IP Protocol in establishing connectivity. Given their features, they have advantages vis-à-vis other access technologies such as DSL [G.992.5], but also challenges, due to being technologies through which all services will be made available over IP, IPv4 or IPv6 (All-IP). Voice is one of them.

2.1 LTE/EPC

In the framework of the 3GPP, the next benchmark in the evolution of radio communication systems is LTE (E-UTRAN).

It consists of a new radio interface, providing more resources through the introduction of new coding technologies and antennas.

At the same time, 3GPP has defined a new entity framework for network access, more simplified and flat, which it calls EPC/SAE. Together they are called EPS (Evolved Packet System) (see Figure 1).

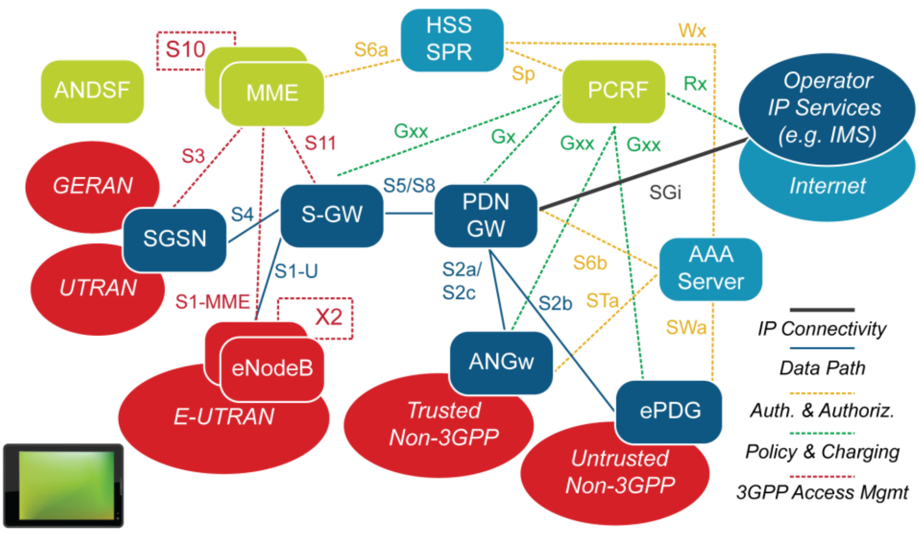

Figure 1: EPC and the different access networks[1]

In the EPC architecture, there are entities that allow access using other technologies that are not only LTE. In addition to 3GPP technologies (GERAN, UTRAN, and E-UTRAN), it is possible to obtain access via WLAN. In this way, and only with recourse to networking technologies, personal discrete (nomadic), service and terminal mobility are possible. To this end, there are specific gateways: ANGw and PDG.

[1] FUSECO Forum 2011, Berlim, Nov/11, Tutorial 1, “Control Platforms and Applications for Next Generation Networks and the Future Internet”, Prof. Dr. Thomas Magedanz

Since IMS is a control architecture that is also defined in the scope of the 3GPP, LTE/EPC presupposes that multimedia services be made available integrating both (see figure). Other services will exist that take advantage of the higher bandwidth and lower delays made available with LTE/EPC technology.

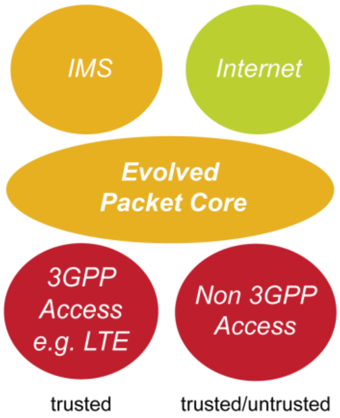

Figure 2: IMS positioning in the EPC

These are the main features of LTE/EPC:

All-IP	Impact on traditional voice and SMS services that will have to be supported on IP (v4 or v6)
Higher bandwidth and lower delay	Increased QoS network level (IP connectivity) Breeding ground for the OTT to bloom.
It incorporates mechanisms for integrating with other networks	Possibility of more efficient network change process (handover) Bird's-eye view of the network, with a single core (EPC)
It incorporates mobility mechanisms, complementing the service level mechanisms	Use of own mechanisms of radio technology and MIP
It will not have initial global geographical coverage	Terminals will have to use other networks (2 and 3G) to access services - necessary complementary mechanisms to guarantee continuity

2.2 ANDSF - Access network discovery and selection function

The ANDSF is an entity of the EPC with the objective of assisting the user’s terminal equipment in the selection of the best network access, for the service conditions that it presents. It holds a database of policies that are delivered to the terminal equipment. With these policies, the terminal takes note of the networks available in the location and how to use them, making it an important element in support for terminal and session mobility situations.

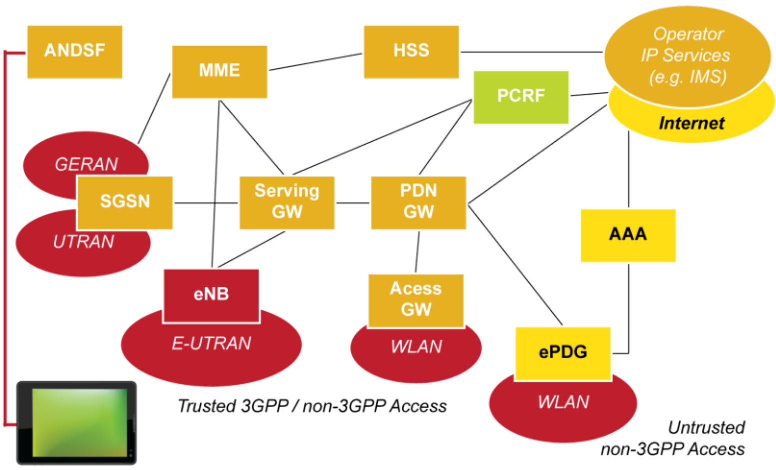

Figure 3: ANDSF in EPC architecture[2]

In its current state of standardisation, ANDSF only has one interface (S14) with terminal equipment, based on OMA DM (Device Management). Thereby, there must also be an entity in the terminals establishing the necessary dialogue with the ANDSF server in order to obtain the policies that apply and make them run internally.

2.2.12.2.1 MME Mobility Management Entity

The MME is a central control entity in the core of the access network (EPC) (see figure). This entity has several functionalities, which are fundamental to the new technology:

- Control of call status
- Management of transport resources
- Authentication
- Paging
- Mobility of nodes
- Roaming

For integration with other networks, the MME has interfaces defined with elements with elements of from other technologies, such as the SGSN and GPRS. For some of the features described below, for example CSFB and SRVCC), it also has interfaces with MSC (supporting entity of 3G voice service - service given in CS - Circuit Switching - mode).

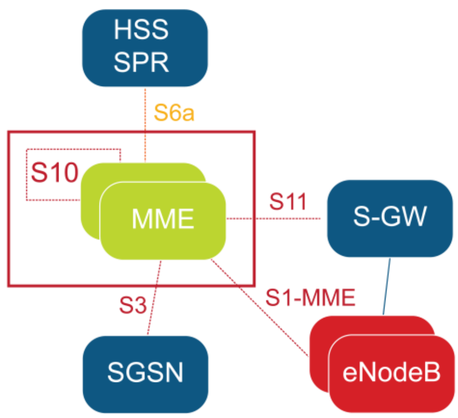

Figure 4: MME in 3GPP3 networks3

[2] FUSECO Forum 2011, Berlim, Nov/11, Tutorial 1, “Control Platforms and Applications for Next Generation Networks and the Future Internet”, Prof. Dr. Thomas Magedanz

2.3 IMS

The IMS, IP Multimedia Sub-system, consists of open and standard modular architecture, defined by the 3GPP [TS 23.228] in Release 5, aiming to efficiently provide multimedia services over an infrastructure providing IP connectivity between the communicating entities. As such, it reuses the IETF defined protocols, including SIP [RFC 3261] and Diameter [RFC 3588].

Initially set for implementation to wireless communications sector, its benefits in a wider application have been perceived and it is currently a centrepiece in the construction of next generation networks (RPG or NGN- Next Generation Networks) where the fixed-mobile convergence comes to boost the creation of new services and business models. In this context, its adoption implies a new approach to the way of doing telecommunications, with infrastructure and organisational changes, to fully take advantage of its benefits.

The core IMS architecture (CN) is developed based on the entities indicated in the following figure:

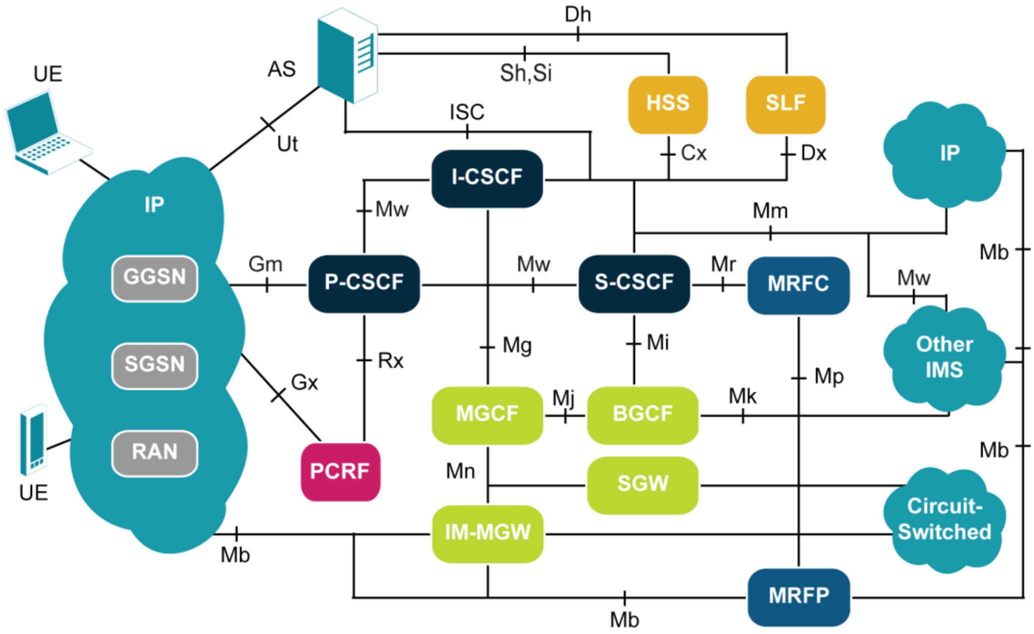

Figure 5: General IMS and points of reference[3]

2.3.12.3.1 IMS Services

One of the great advantages of the IMS is its capacity and flexibility to provide a dynamic set of services.

The services available through the IMS are run on Application Servers (AS). These are used in the IMS architecture for the provision of value-added IM services and may reside on the network of the subscriber or at another location (Third-parties, with which there is a network connection with the appropriate security levels) but accessed from the first. A principle underlying the IMS is that services should be implemented in the Home Network, i.e. in the subscriber's own network. This means that for a user in a visited network (roaming), the execution of the services that are subscribed will be invoked on the network to which the service subscription belongs to.

[3] http://www.hill2dot0.com/wiki/images/thumb/4/43/IMS_Architecture_RefPts.jpg/800px-IMS_Architecture_RefPts.jpg

The existence of multiple AS is normal in a network, with each one specialising in a particular service, a fact that is reflected in the logic performed by them. Its implementation can be based on various technologies like Java, SIP servlets or SIP CGI (Common Gateway Interface). An AS, according to the service performed, influences and has impact on the SIP session.

In a basic call, the AS are not necessarily involved, but in situations where you want to perform more complex functions, its service logic must be invoked, acting on the signal generated by subscribers. To do this, they must be involved in the signalling process, on the initiator's and/or terminator's side of the session. This happens through the ISC (IMS Service Control), established between the AS and the S-CSCF of the domain, constituting an interface based on the Session Initiation Protocol (SIP).

It is the S-CSCF that performs the decision-making process of whether or not to involve a given AS in a call, this decision being based on the Filter Criteria received from the HSS, via the Cx interface, during the process of registration of a subscriber (initial FC) or being subsequently defined (subsequent FC). These filtering criteria are defined in the profile of the user, provisioned in the HSS, and are defined at the time that the user is created and changed according to the subscribed services. In turn, the AS can also communicate with the HSS through Sh and Si interfaces. More than one AS may be involved in the same session, on the originator's and/or terminator's side.

3. Types and level of mobility execution

The guarantee of mobility, session continuity, and the universality of the defined solutions, are important differentiators of incumbent operators vis-à-vis Over-the-Top (OTT) service providers. Operators are the owners of the infrastructure, including 3GPP technology (2/3/4G) and WLAN, and can differentiate between services and guarantee quality, according to the value of the service.

Various types of mobility can be identified:

1. **Personal:** Being globally communicable (same person, different networks/terminals); being able to access your services anywhere, anytime, using any network/terminal.
2. **Terminal:** Possibility that a terminal has of changing a point of connection to the network, using the same or different network interfaces of the same or different technologies (even different networks/terminals).
3. **Session:** Ability to move a session between different terminals (same session/different terminals).
4. **Service:** to have access to the subscribed services and its features regardless of the provider used, e.g. in roaming (same services/different networks).

To respond to these types of mobility, different mechanisms have been defined, acting at different levels of the protocol stack, and can be used alone or together:

1. **Link-layer**: The use of the technology’s own mechanisms, especially important in wireless communication networks (2G/GSM, 3G/UMTS, 4G/LTE).
2. **Network-layer**: Mobility made at the IP level, achieved with recourse to MIP (Mobile IP), and variants (MIPv6, PMIPv6, DSMIP).
3. **Application-layer**: Use of application signalling mechanisms, in this context via the SIP protocol and IMS control core, and extensions (for example, VCC or SRVCC).

3.1 Link-layer Mobility

In general, all wireless access network communication technologies (radio), include terminal mobility solutions. While these networks are typically organised by geographical areas with a central entity where the mobile terminals are switched on, with these mechanisms the geographical terminal mobility field widens beyond the radio coverage of such entities, considered individually. It is thus that the terminals move throughout the geographical area covered by this technology, changing the point of connection to the network, always looking for the best quality.

The embedded mechanisms in the technologies work in a way that changes in the network connection point are not perceived by the upper layers (IP level). In the case of 2G technology, this also includes the Circuit Switching (CS) components, supporting voice and SMS services.

Examples of supported mobility intra-technologies are the IEEE 802.16, IEEE 802.11, GPRS, UMTS and LTE/EPC, which have their own mechanisms.

3.2 Mobile IP (MIP)

The MIP is used to solve the problem of the mobility of terminals connected to IP networks, solving the problem at the network level. In IP networks, changing the connection point between IP sub-networks implies an IP change which interrupts the active connection and requires that the new IP is made known to its corresponding IP so that the node remains connected. The use of the MIP allows a terminal to continue to be identified by a single IP address (Home Address-HoA) of the original network (Home Network), even when it moves between other networks (Foreign Networks). It requires the definition of new entities, HA (Home Agent, in both IPv4 and IPv6) and FA (Foreign Agent, only in IPv4). Basing its functioning on mechanisms solely at the network level (IP), it is transparent to transport protocols and, consequently, to applications. However, this mechanism ensures the nomadism of the terminal but not solid mobility, since the network change process, which involves assigning to each visited network a temporary IP (CoA-Care of Address) and its registration in the HA, can be time consuming and involve disruption at the application level (packet loss, with eventual connection loss). The MIP already exists in the IPv4 [RFC 5944] and is improved in IPv6 [RFC 6275] by taking advantage of some special features of the Protocol, such as the use of router advertisements, simplifying its operation.

The MIP can be used when the different networks where the terminals are connected are of the same technology (intra-technology or horizontal handover) or of different technologies (inter-technology or vertical handover).

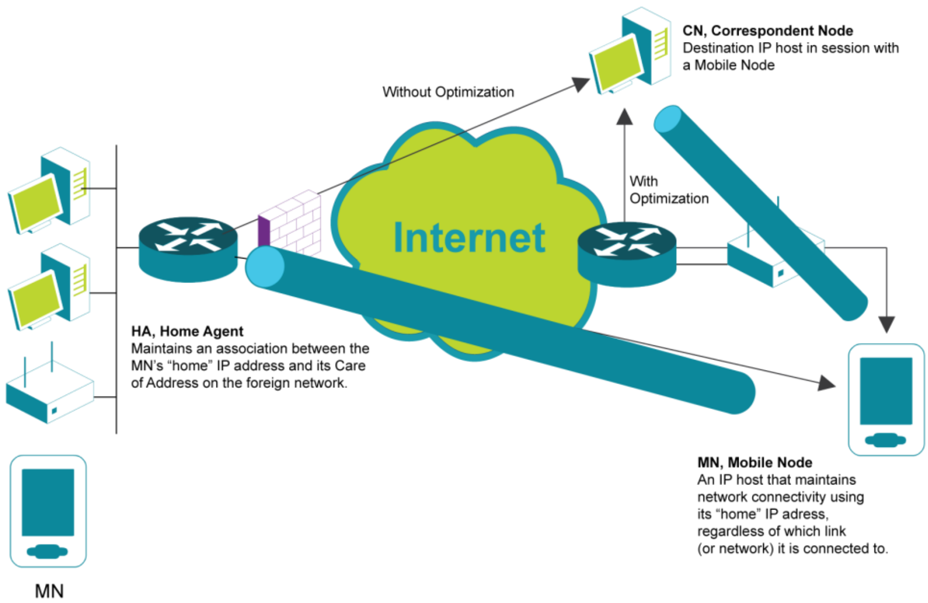

Figure 6: Mobile IP (MIP)

There are variants of the protocol such as PMIPv6 [RFC 5213], with mobility control to be made by the network, and DSMIPv6 [RFC 5555], for dual-stack nodes (nodes with concurrent IPv4 and IPv6 connectivity).

3.3 SIP Mobility

With the SIP protocol [RFC 3261], session mobility is possible. Using the method REFER [RFC 3515], it is possible to transfer one of the terminations of an SIP session in progress.

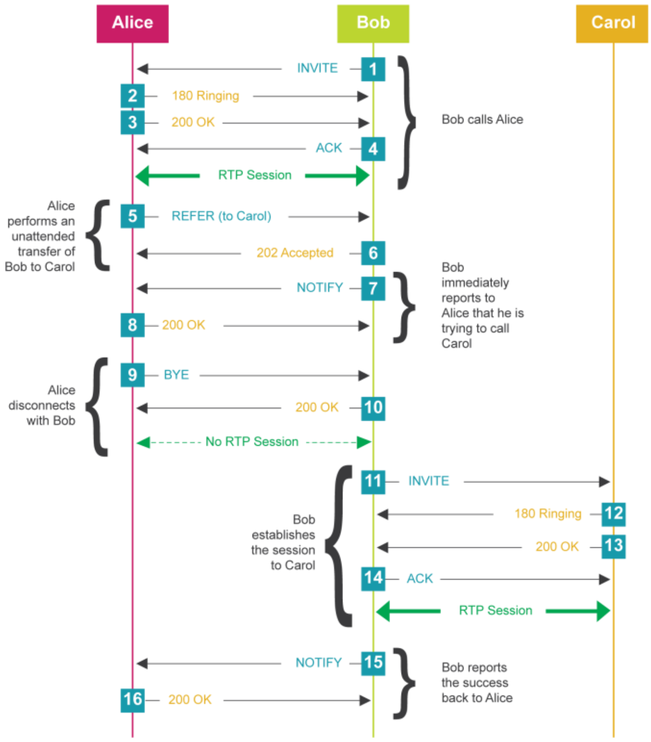

Figure 7: Call transfer [Tech Invite]

The use of reINVITE during a session in progress, can also be done.

Other more complex processes, also using the SIP protocol, are possible. RFC 5631 describes methods of transferring media components between devices in ongoing sessions.

Terminal mobility is achieved through its process of registration. Before it is possible to perform any action to establish a session, usually initiated by the INVITE method, the terminal (more precisely the UA-User Agent - resident in the terminal) has to register on the SIP domain to which it belongs, by using the REGISTER method. Whenever the terminal changes IP, it must repeat the registration process in order to create a new correspondence record between the new IP address and the SIP identifier for which it is known at the level of service. Whenever such a change occurs, active sessions end. Considering that the SIP identifier belongs to an individual, we can actually identify this type of mobility as being personal mobility.

4. Cs and PS domain convergence aspects

4.1 CSFB - Circuit Switch FallBack

Services of the CS (Circuit Switch) domain in wireless 3GPP networks, consist of services that can currently be given by GSM/UMTS networks. They specifically consist of basic and additional voice, SMS, USSD services, among others. The EPS will allow access to much richer services, but in the early stages of deployment this may not be the case, and should be used primarily to provide faster access to data services. In addition, the geographic coverage of the LTE may not be complete, with some areas having only 2G/3G coverage.

In this context, the CSFB is introduced in Release 8 of the 3GPP [TS 23.272] to allow a mobile terminal that is registered in an EPS network to use services in the CS domain, switching its radio interface from EUTRAN to another RAT (GERAN or UTRAN). This scenario is especially important at an early stage of operation of the LTE, where technology is used only for data communication or while the geographical coverage is lower than that of CS technologies.

In this way, this terminal can take advantage of the best quality for data communication by switching to a CS domain only when available services are required, for example to receive or establish a voice call, returning to EPS as soon as the use of these services is concluded. Any data services that are being used in the EPS are suspended and resumed in the return to this network. The CSFB also defines a way of a terminal that is connected to EPS being able to send SMS, without getting off of this network. Thereby, it is not about supporting mobility or session continuity but rather about ensuring that a given service (voice) is provided to the user by moving the connection point between LTE/EPC and another legacy technology, with support for the required services.

In the following figure (Figure 8) is presented the reference architecture for CSFB, requiring the intervention of the entities represented here, in particular the UE, MME, SGSN and MSC. A new interface, SGs, derived from the Gs, appears, uniting the MME to MSC, used to manage the fallback. This interface will allow the UE to be notified that it should switch to receive a call. Also, a new S3 interface appears between MME and SGSN.

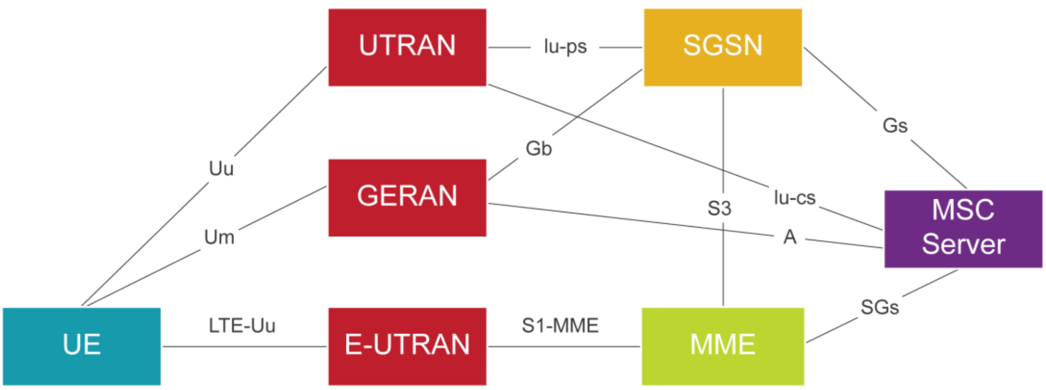

Figure 8: Reference architecture for CSFB

To use the services of the CS domain, for example, to receive or establish a voice call, the UE is registered in the CS network, via EPS. During this process, the UE is informed if in that location it is possible to carry out the CSFB.

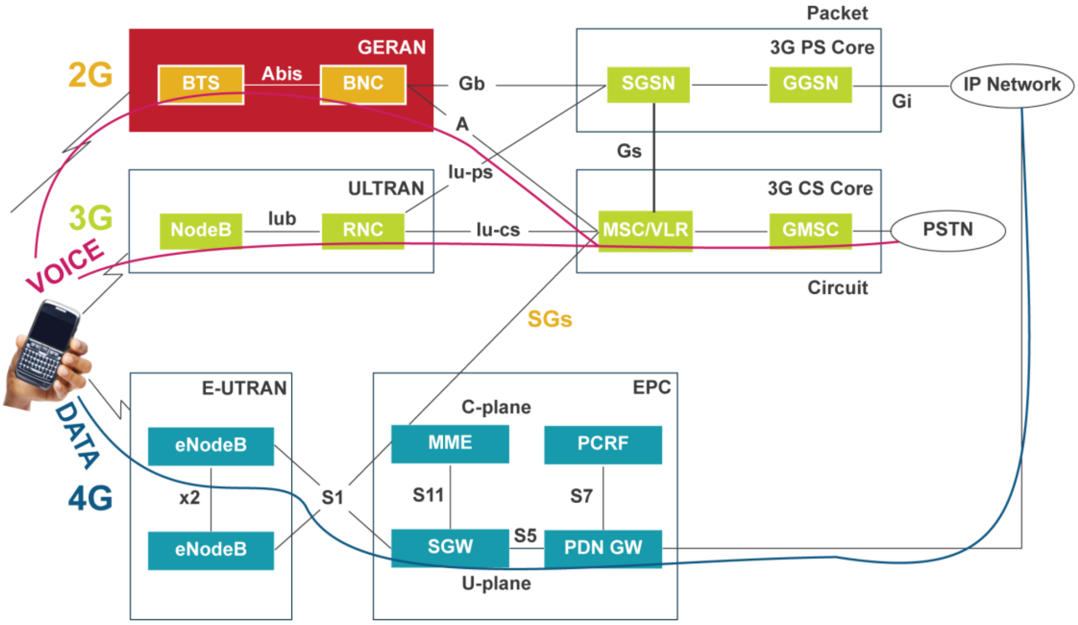

Figure 9: CSFB Operation[4]

In this way, if there is a data session in the LTE network, it is suspended, the terminal changes access network (RAT), makes the voice call and, after this, returns to the LTE network, resuming the data session. With this approach, there are several problems:

- Additional time to make the call
- Network coverage requirements
- Loss of data connection during voice calls, with impacts on the applications that use them
- Implementation costs and feasibility of the solution
- Negative impacts on the current and future business models for LTE
- Limiting model for the implementation of new types of voice applications
- Integration problems with femtocells

For these reasons, many operators are considering putting the CSFB in operation, due to the costs involved and to the poor perception of the service that customers will have waiting for other solutions, with LTE, in the meantime, being used only in data terminals (PCs, Tablets, other). Additionally, it capitalises on the investment that was made or will be made in IMS.

Contents:

- Applicable to EPC and 2G/3G networks
- Process started in the LTE/EPC network
- Allows that services requiring CS not fail to materialise
- Control in the PS and CS
- Requires interface (SGs) between PS (MME) and CS (MSC)

[4]http://www.nil.si/ipcorner/VoLTEImplementations/

4.2 ICS – IMS Centralised Services

ICS, defined by the 3GPP in Release 8 [TS 23.292], provides centralised control of services provided on an IMS core (IM-CN) [TS 24.237], at various types of access, namely fixed and mobile. With the ICS, all services are provided by Application Servers (AS) IMS, regardless of access (bearers) used (CS or PS). If the access network is PS and supported to establish the voice component of an IMS service bidirectionally, the IMS control (IM-CN) treats it as a normal IMS session. When using a CS network or a PS network, but without capacity to bidirectionally withstand voice components, the CS network is used to establish the necessary access circuits to the passage of the media, under the control of the IM-CN.

It requires the existence of an AS for SCC (Service Centralisation and Continuity). This is inserted into the SIP path through source and termination iFC:

- It must be the first AS in the sequence of source iFC;
- It must be the last in the sequence of termination iFC.

The overall architecture is presented in the following figure, introducing the following elements:

- AS SCC, offering the functionality required, located in the UE network origin;
- MSC servers extensions to integrate with the IMS core and to control MGW for interconnection of the PS and CS domains;
- Extensions to the terminals supported by ICS, to be able to request ICS services.

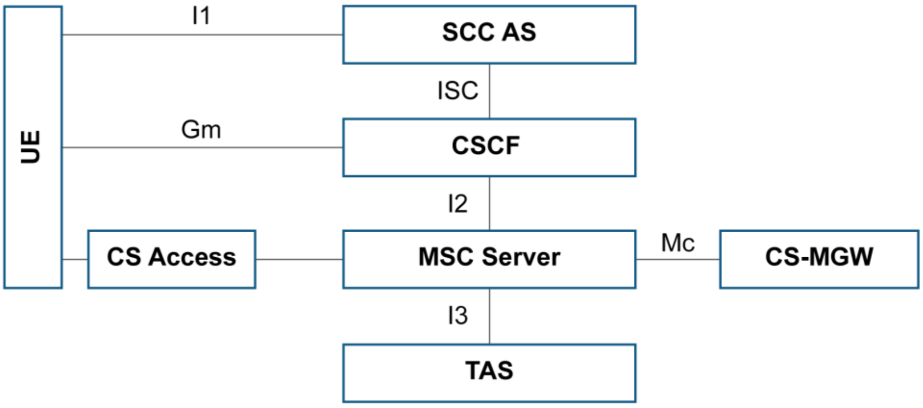

Figure 10: Reference architecture for the ICS [TS 23,292]

The registration of the UE in the IMS core, when located in the CS domain, is made by the MSC. The same is true with flagging. It is the MSC that performs the translation between CS and PS (SIP) signalling. Interface I1 connects the UE directly to the AS SCC. Its use is described in [TS 24.294].

Summary:

- Applicable to any PS or CS technology, with the mobile terminal to maintain access to services either via PS or CS control interface (translated in the MSC server)
- It allows sessions that have CS and PS components to be established in the IMS core
- It allows sessions that would be totally PS, but that are unable to perform, to materialise with recourse to CS for the medium but with PS control
- It doesn't support mobility with active sessions (service continuity)

It does not allow merging and separation of components of a multimedia session

5. Continuity of Service

In a scenario of a single technology, and in the context of wireless access networks ("mobile networks"), terminal mobility is normally guaranteed by the access technology used or by the network level (IP). In this situation, services like RCS or VoLTE can be used, continuously and in motion.

However, with the coexistence of different technologies, it is natural that there be a need to establish mechanisms for mobility between these networks. This need can arise from the existence of different areas of coverage of the different technologies, from the search for the best price or from the needs of the network itself in alleviating load in a technology, moving the terminal to another one with more available resources (offload). In these cases, there will be change of RAT, which, from the level of service, consists of session mobility between different network access points.

The inclusion of additional mechanisms, such as the SCC (Service Centralisation and Continuity), allows mobility of the login component between different terminals, under the control of a same user. These mechanisms assume that services are provided using an IMS control centre and that there are Application Servers (ASs) where control signal anchoring of multimedia sessions is done.

5.1 SRVCC - Single Radio Voice Call Continuity

Specified in Release 8 of the 3GPP [TS 23,216], SRVCC is the evolution of the VCC (Voice Call Continuity) and SCC (Service Centralisation and Continuity). It is a feature that allows a voice IMS session to have continuity when the terminal has only one active radio interface, forcing it to only be active in one RAT at a time (E-UTRAN, UTRAN, GERAN and 1xRTT). So, when one moves out of the LTE coverage area, for example, it allows for the continuity of service, running a handover to a legacy CS network (GSM/UMTS or CDMA 1 x). This functionality is particularly interesting for the beginning of the operation of the LTE/EPC, with the provision of voice service with recourse to VoLTE and when the geographic coverage is lower than that of legacy technologies. In these circumstances, it will be necessary to carry out the handover between LTE/EPC and those technologies, with active sessions and in situations where the terminals cannot have more than one active radio interface at any time. Thus, supporting services such as VoLTE, and for the operator that already possesses IMS, SRVCC allows to extend the scope of coverage of its voice services. The figure below shows the reference architecture for the RAT EUTRAN and UTRAN/GERAN situation.

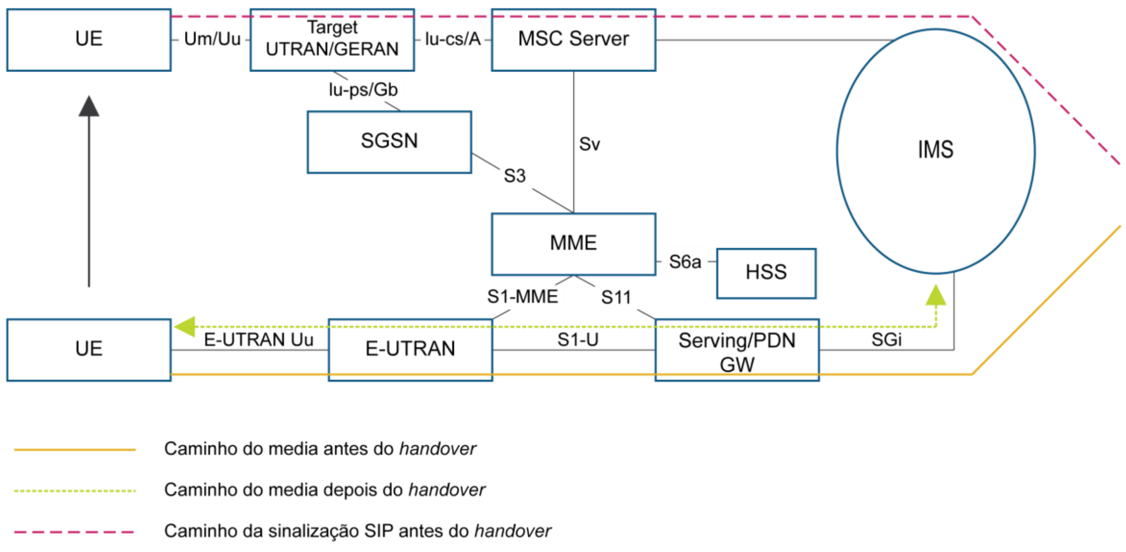

Figure 11: SRVCC Architecture for E-UTRAN network transition to 3GPP UTRAN/GERAN [TS 23.216]

The SRVCC is built on ICS which requires an SCC AS, functioning as B2BUA, and being the signalling anchor point for all calls subject to SRVCC functionality, run by an SRVCC AS, and for execution of its transfer between LTE and WCDMA/GSM domains.

In order for that to work, it is necessary to understand the functionality and terminals that implement some functional requirements. During the process of connecting to the network, they must indicate having support for SRVCC.

In addition to the AS of the SRVCC and of compatible terminal equipment, extensions to the MSC are necessary, which should support the Sv interface for connection to MME, which plays a central role. An MSC enriched to endure SRVCC functionality can also be enriched for ICS functionality.

In the following figure is exemplified the EUTRAN to GERAN/UTRAN change process. It should be noted that before the implementation of the SRVCC functionality, the PS domain must separate the voice components from the data (bearer splitting). These components will be treated independently.

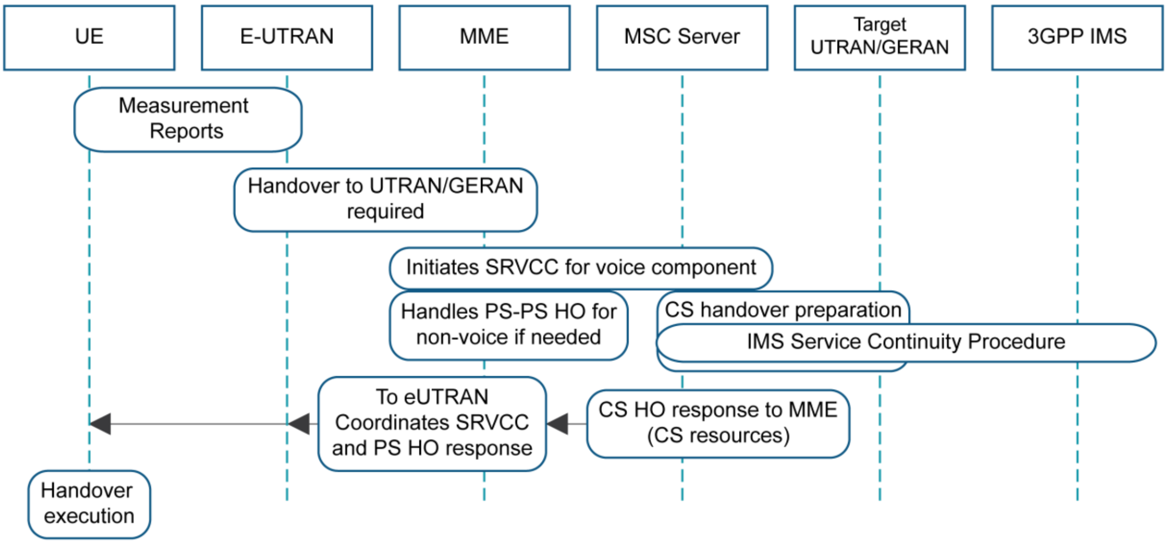

Figure 12: Summary of RAT process change (E-Utran-->GERAN/UTRAN)

In the case of a single radio, handover must be prepared in advance in order to minimise the impact. The MME receives a request for handover from the EUTRAN, indicating that this should be treated as SRVCC. Via the Sv interface, MME contacts MSC, which in turn contacts the IMS core, initiating the session transfer process, and coordinating it with the CS to the target UTRAN/GEGAN cell. The process ends with a statement to the UE that should proceed to the handover, changing RAT.

The existence of PS components that are not voice must also be treated, being supported by MME [TS 23,401].

The SRVCC solution is shown to be the long-term solution, because it uses the advantages of the IMS and adds the functionality of the other features of session continuity and mobility. However, implementation costs can be high, so if the introduction of voice service in the LTE/EPC network is done after this network already has significant geographical coverage, this instrument may also not be able to be used.

Summary:

- The SRVCC consists in the evolution of the previous VCC and SCC, being a long-term solution
- It can be coordinated with other mechanism such as the ICS
- Executing it requires changes in existing network elements (MSC and MME)

6. Conclusions

The EPC is an evolution in wireless network infrastructures, adding other networks that are not only LTE. In addition to terminal mobility, management flexibility and session continuity are also important dimensions. The mechanisms for ensuring session continuity and mobility are diverse, acting at different levels of the protocol stack. Some concepts on continuity of services associated or not associated to terminal mobility have evolved. Currently, the CSFB and SRVCC are among the solutions that operators should consider placing in the roadmap of the evolution their networks.

7. References

- 3GPP

[TS 23.216] TS 23.216, Single Radio Voice Call Continuity (SRVCC)

[TS 23.228] TS 23.228, IP Multimedia Subsystem (IMS)

[TS 23.292] TS 23.292, IP Multimedia Subsystem (IMS) Centralised Services

[TS 23.401] TS 23.401, General Packet Radio Service (GPRS) enhancements for Evolved Universal Terrestrial Radio Access Network (E-UTRAN) access

- IETF

[RFC 3261] RFC 3261, SIP: Session Initiation Protocol

[RFC 3515] RFC 3515, The Session Initiation Protocol (SIP) Refer Method

[RFC 3588] RFC 3588, Diameter Base Protocol

[RFC 5213] RFC 5213, Proxy Mobile IPv6

[RFC 5555] RFC 5555, Mobile IPv6 Support for Dual Stack Hosts and Routers

[RFC 5631] RFC 5631, Session Initiation Protocol (SIP) Session Mobility

[RFC 5944] RFC 5944, IP Mobility Support for IPv4, Revised

[RFC 6275] RFC 6275, Mobility Support in IPv6

- ITU-T

[G.992.5] G.992.5, Asymmetric digital subscriber line (ADSL) transceivers – Extended bandwidth ADSL2 (ADSL2plus)

- URL

[TechInvite] http://www.tech-invite.com/

8. Acronyms

3GPP	3rd Generation Partnership Project
ABC	Allways Best Connected
ANDSF	Access Network Discovery and Selection Function
ANGw	Access Network Gateway
AS	Application Server
CGI	Common Gateway Interface
CS	Circuit Switching
CSCF	Call Session Control Function
CSFB	Circuit Switched FallBack
DSL	Digital Subscriber Line
DSMIP	Dual-Stack MIP
EPC	Evolved Packet Core
EPS	Evolved Packet System
E-UTRAN	Evolved UTRAN
FA	Foreign Agent
FC	Filter Criteria
FMC	Fixed-Mobile Convergence
GERAN	GSM EDGE Radio Access Network
GPRS	General Packet Radio Service
GSM	Global System for Mobile Communications
HA	Home Agent
HSS	Home Subscriber Server
ICS	IMS Centralized Services
IMS	IP Multimedia Subsystem
IP	Internet Protocol
ISC	IMS Service Control
ITU-T	International Tecommunication Union - Telecommunication Standardization Sector
LTE	Long Term Evolution
MIP	Mobile IP
MME	Mobility Management Entity
MSC	Mobile Switching Center
NGN	Next Generation Network
PDG	Packet Data Gateway
PMIP	Proxy-MIP
PS	Packet Switch

RAT	Radio Access Technology
RFC	Request For Comments
SCC	Service Centralization and Continuity
SGSN	Serving GPRS Support Node
SIP	Session Iniciation Protocol
SMS	Short Message Service
SRVCC	Single Radio Voice Call Continuity
TS	Technical Specification
UA	User Agent
UMTS	Universal Mobile Telecommunications System
UTRAN	Universal Terrestrial Radio Access Network
VCC	Voice Call Continuity
VoLTE	Voice over LTE
WLAN	Wireless LAN

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