8

Voice over LTE (VoLTE)

In the mobile business, the telephony service has been the major source of revenue for most mobile operators. Even with the proliferation of smartphones, the voice service is still considered important by users. For the success of LTE, it is of crucial importance to maintain support for the voice service over the LTE network.

8.1 Voice Solutions for LTE

8.1.1 Ultimate Voice Solution

The open architecture of the Internet Protocol (IP) created the flexibility that has enabled a thriving multitude of multimedia services to be available over the Internet. To integrate mobile access with the IP multimedia services, 3GPP has defined an architectural framework, called the IP Multimedia Subsystem (IMS), to deliver IP multimedia services over 3GPP systems. The success of the IMS in 3GPP motivated other organizations in the same industry, such as 3GPP2, to adopt the IMS as the architectural framework for IP multimedia services. The IMS also supports integration with the wired network, creating the potential for fixed–mobile convergence.

The mobile industry has been motivated by the success of the IMS standard, and the richness and flexibility of the IMS architecture provides a reliable framework to enable fruitful multimedia services including a voice service with guaranteed quality of service. The IMS is considered by most operators to be the ultimate architecture to provide a voice service over the LTE network. Figure 8.1 illustrates the concept of a voice service based on the IMS architecture deployed over the LTE network. For the sake of convenience, a voice service based on the IMS architecture is referred to as "IMS VoIP" hereafter.

To minimize implementation options in IMS VoIP, the industry collaborated to define a common solution for IMS VoIP by organizing the One Voice Initiative in November 2009. The One Voice Initiative produced a "One Voice; Voice over IMS profile" that captures a minimum mandatory set of IMS and other features for UEs and networks to support for voice and other services, for example, SMS. The work of One Voice was strongly supported by the GSM Association (GSMA) and taken over by the "GSMA VoLTE Initiative" that was

Radio Protocols for LTE and LTE-Advanced, First Edition.
SeungJune Yi, SungDuck Chun, YoungDae Lee, SungJun Park and SungHoon Jung.

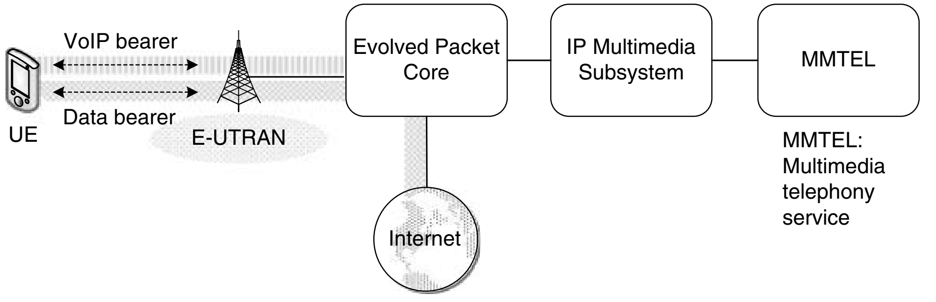

Figure 8.1 Concept of IMS VoIP over the LTE network

established to promote a common voice solution over LTE. The GSMA VoLTE Initiative published an official document (IR.92) which defines the IMS profile for voice and SMS.

8.1.2 Interim Voice Solutions

Given today's mega trend of IP convergence, one would expect that, in years to come, IMS VoIP will be universally available over both the wired network and the wireless network including LTE.

Even though the LTE system (E-UTRAN and EPC) supports only packet-switched (PS) IP services and the IMS VoIP solution has been agreed as the industry-wide solution for the voice service, the industry cannot rely solely on the IMS VoIP solution.

There are legacy networks and legacy mobile devices that can only work on the legacy networks with legacy solutions (i.e., CS voice telephony), and the advent of LTE cannot replace the entire legacy system on a grand scale on the same day. Recalling the enormous investment already made in the legacy networks, some operators may be reluctant to evolve their legacy networks towards an LTE network in a hurry. In general, different operators will have quite different network evolution strategies, which will result in a heterogeneous environment where the LTE network will coexist with legacy networks for quite some time. Even though it is an essential requirement that service continuity between the networks and the quality of service for voice telephony should be guaranteed, the fundamental difference between the LTE network and legacy networks makes satisfying this essential requirement a challenge.

The roaming scenario adds further complexity to the support for only IMS VoIP on the track of network evolution towards LTE. For example, while supporting only IMS VoIP may work on the home LTE network, it may not work on a visited network where IMS VoIP is not supported or only a legacy network is available.

As per IMS deployment, it may take a non-trivial amount of time for IMS infrastructure and IMS-supporting UEs to be widely available in the target market. Nonetheless, LTE deployment should not be delayed until IMS deployment is fully completed over all service areas, as LTE networks are likely to be used to provide only data services in the early phase of the deployment. As a result, a voice service other than the IMS VoIP should be provided over legacy networks in the initial LTE deployment phase.

Even though IMS infrastructure and/or UEs capable of IMS VoIP are widely available, the LTE network of one operator may cover only limited geographical areas in the initial phase of LTE deployment, while legacy 2G/3G coverage will provide access in all areas covered by the operator. Because the voice service should not be suspended as a result of the unavailability of LTE coverage, the issue of how IMS VoIP should be handled at the border area of LTE coverage needs to be resolved.

In short, it should be understood that supporting only a simple voice solution like IMS VoIP is insufficient in practical terms, and interim solutions in addition to the IMS VoIP service will need to be used until the IMS VoIP solution becomes mature on a global scale.

The mobile industry and standard groups considered the following interim solutions:

- Circuit-Switched FallBack (CSFB);
- Voice over LTE via Generic Access (VoLGA).

Figure 8.2 illustrates CSFB and VoLGA. CSFB is the standardized solution that provides a voice service over the existing 2G/3G circuit-switched (CS) network. The principle of CSFB is that the UE is normally camping on LTE, but it is moved to a 2G/3G CS network when a voice service is to be initiated. CSFB is widely supported by many operators as it requires minimum changes to legacy 2G/3G networks, and CSFB fits well with the operators' strategy that 2G/3G systems will provide voice services and LTE systems will provide data services only in the early phase of LTE deployment.

Another proposed solution was VoLGA, which extends the existing Generic Access Network (GAN) to support voice over the LTE network. GAN was designed originally to provide mobile services to the UE via supporting both the 3GPP radio interface and the Wi-Fi radio interface. A GAN gateway provides a secure connection to deliver circuit-switched services to a subscriber via a Wi-Fi access network. The key concept of VoLGA is to replace the Wi-Fi access network with an LTE access network in the GAN architecture. VoLGA did not get much support from the mobile industry and standard groups.

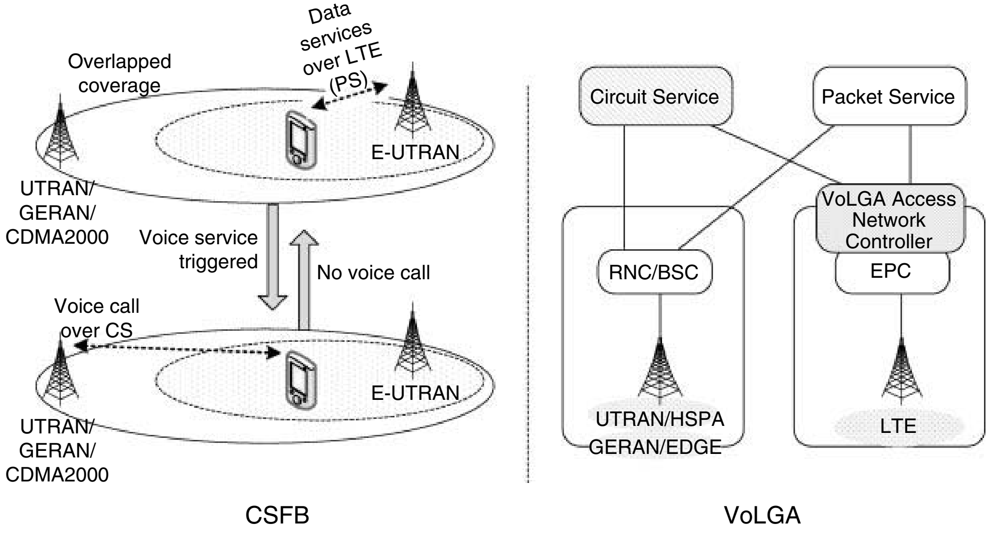

Figure 8.2 CSFB and VoLGA

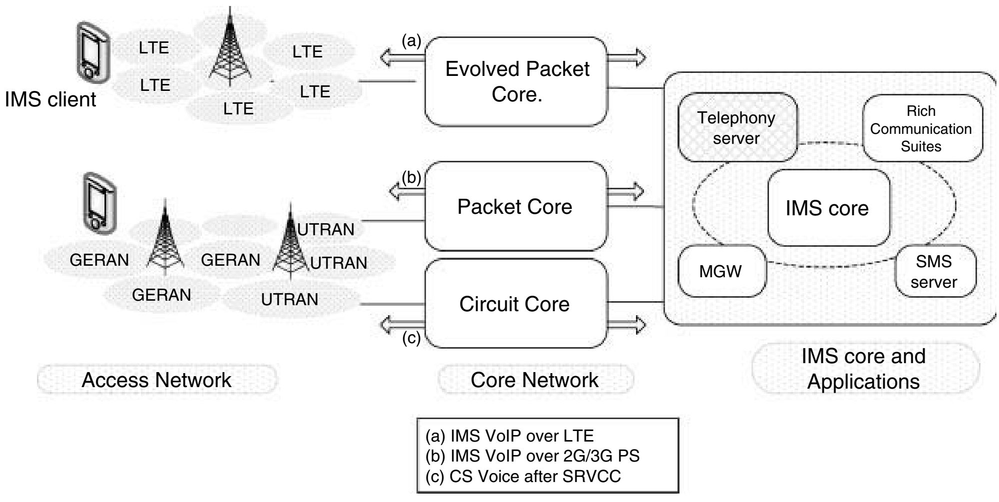

Figure 8.3 Concept of IMS VoIP

8.2 IMS VoIP

The IP Multimedia Subsystem (IMS) is a standardized architectural framework to provide IP-based services. The IMS supports IP multimedia applications including voice, messaging, video, conferencing, and blended multimedia services. There are advantages to using the IMS framework for voice services:

- global roaming with fewer interoperability problems is ensured;
- full performance of the LTE during voice calls is sustained;
- high-definition voice services can be offered readily;
- blended services with video/conferencing/data can be enabled gracefully.

The IMS requires new IMS-specific network elements as part of the dedicated core network architecture. Figure 8.3 illustrates the concept of IMS VoIP over 3GPP networks. Note that IMS VoIP also accommodates 3GPP2 networks, which are not shown in the figure for simplicity.

8.2.1 IMS Profile

Even though the IMS can enable rich multimedia services including a voice service on an IP-based network, the rollout of the IMS has been delayed by many mobile operators. The main reason for the delay in IMS deployment is that the architecture of the IMS is very complex as a result of many architectural options depending on target services and the access/core network in question. If different IMS-based voice solutions were implemented by different operators, it would not be possible to avoid the risk of LTE market fragmentation and interoperability issues. This would also result in a situation where the chance of roaming for subscribers could be quite limited depending on the solution adopted by the operator.

8.2.1.1 One Voice Initiative

Faced with the practical threat of fragmented IMS deployment, the industry recognized the importance of an industry-wide common solution to promote a viable ecosystem for an IMS-based voice service and to enjoy the benefits of economies of scale. The principle of introducing a common standard solution for IMS VoIP is also important to ensure seamless global roaming experiences for subscribers. Motivated by the necessity of a global working standard solution, the industry established the "One Voice" Initiative in November 2009.

The One Voice Initiative adopted the IMS architecture defined by 3GPP for provisioning a voice service and SMS over LTE networks. With the aims of accelerating the deployment of a common solution for an IMS-based VoIP service and avoiding complexity in the initial phase of IMS deployment, the One Voice Initiative produced a "One Voice Profile". As the word "One" in the name suggests, the objective was to define a single profile for an IMS VoIP service rather than multiples to reduce IOT efforts and implementation costs by defining a de facto solution. The "One Voice Profile" defined a minimum mandatory set of features that should be implemented by UEs and networks to support target IMS services over LTE based on 3GPP specifications.

8.2.1.2 IMS Profile for Voice and SMS

In February 2010, the GSMA VoLTE Initiative was established with the backing of many companies and organizations to promote voice and messaging services over LTE. The companies endorsing the GSMA VoLTE Initiative were supporters of a common IMS-based voice solution for the next generation of mobile networks. The work of the One Voice Initiative was adopted gracefully by the GSMA VoLTE Initiative as the baseline for further work.

The GSMA published the "IMS Profile for Voice and SMS", which defines a profile specifying a minimum mandatory set of features based on 3GPP specifications for UEs and networks. The IMS Profile is intended to establish a clean ecosystem for IMS-based telephony services over LTE radio access, ensuring high quality and interoperability. Figure 8.4 shows the protocol stacks for the "IMS Profile for Voice and SMS".

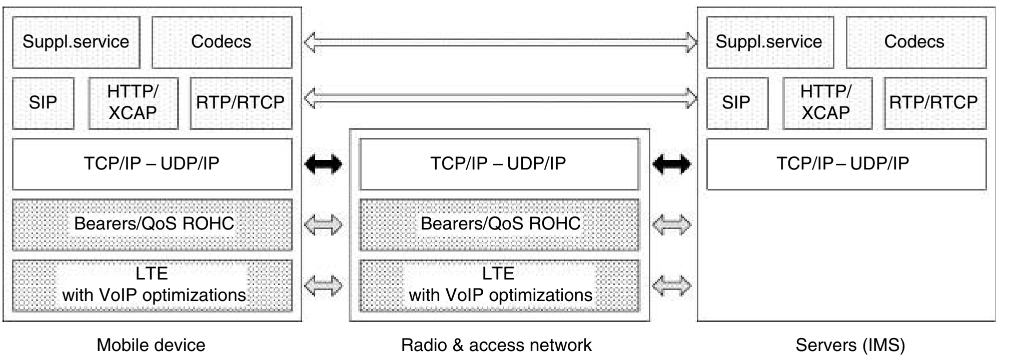

Figure 8.4 Protocol stack for IMS Profile for Voice

The key components of the “IMS Profile for Voice and SMS” are summarized as follows:

- IMS core network with Telephony Application Server (TAS);
- IMS client features at the UE;
- support of Single Radio Voice Call Continuity (SRVCC) for IMS VoIP coverage extension;
- AS features for VoIP support.

The generic IMS functions required for IMS VoIP and IMS VoIP call procedures are beyond the scope of this book.

SRVCC enables the transfer of an IMS voice session from PS to CS. SRVCC is useful for seamless provision of a voice service when IMS voice service delivery over PS is not possible due to PS coverage limitations.

Several AS features help to optimize support for VoIP features, including ROHC, TTI bundling, and SPS.

8.2.1.3 Bearer Characteristics for “IMS Profile for Voice and SMS”

According to the profile, the EPS bearer for the voice service is configured as a Guaranteed Bit Rate (GBR) bearer with a QoS Class Identifier (QCI) value set to 1. The radio bearer for the voice service is configured with UM RLC to reduce overhead while tolerating a small loss of voice data.

8.2.2 Single Radio Voice Call Continuity (SRVCC)

SRVCC provides continuity of a voice call from IMS VoIP over a PS network to CS voice over a CS network for a UE supporting transmission/reception on only one radio at a time. SRVCC is useful for operators with a packet-switched network that covers a limited area and a circuit-switched network that covers the entire service area of the operator. One of the target scenarios for SRVCC is a UE configured with a voice bearer in LTE losing LTE coverage and entering CS coverage. To ensure continuity of the voice call outside of the LTE coverage in this case, the flow of voice data anchored in IMS is transferred from the LTE network to the CS network. SRVCC allows operators to upgrade their 2G/3G networks gradually to LTE networks rather than having to replace their whole networks with LTE all at once.

SRVCC is applicable for voice call continuity from any PS network (e.g., E-UTRAN or HSPA) to any CS network (e.g., GERAN, UTRAN, or CDMA2000). Figure 8.5 shows the network architecture for SRVCC from E-UTRAN to UTRAN/GERAN. The SRVCC procedure is based on the session transfer function defined in [1] for the IMS part, Sv for the interconnection between PS and CS, and the inter-RAT mobility procedure defined in [2] for the access stratum part. The handover from PS voice to CS voice is enabled by utilizing the SRVCC PS to CS procedure via the Sv interface between the MME and the MSC server.

In Figure 8.6, relocation of the voice bearer from a PS network to a CS network through SRVCC is illustrated.

SRVCC from E-UTRAN to UTRAN/GERAN is supported from Release 8. Support for voice continuity in the reverse direction – that is, SRVCC from UTRAN/GERAN to E-UTRAN – will be supported from Release 11.

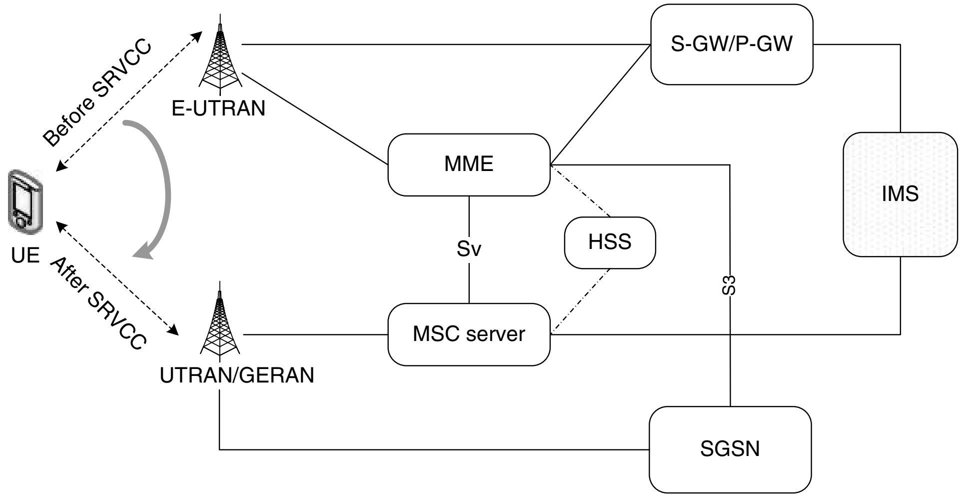

Figure 8.5 Network architecture for SRVCC. Reproduced by permission of 3GPP, © 2011. 3GPP™ TSs and TRs are the property of ARIB, ATIS, CCSA, ETSI, TTA and TTC who jointly own the copyright in them

Figure 8.7 shows a simplified illustration of the SRVCC procedure from E-UTRAN to UTRAN. A UE capable of SRVCC indicates its SRVCC capability as part of the UE capability information during an NAS procedure such as the Attach procedure or the Tracking Area Update procedure. If both the UE and the MME support SRVCC, the MME indicates support for SRVCC to the eNB during the initial context setup over S1 for the UE.

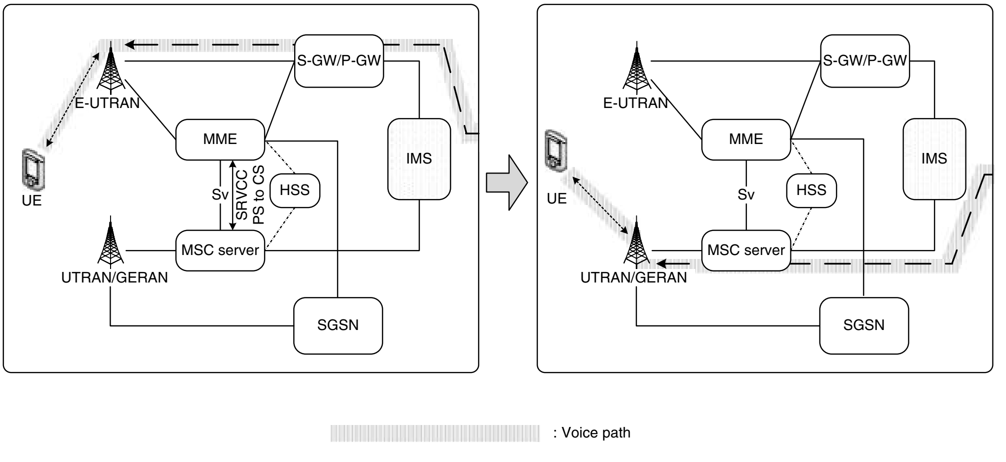

Figure 8.6 Voice path before and after SRVCC. Reproduced by permission of 3GPP, © 2011. 3GPP™ TSs and TRs are the property of ARIB, ATIS, CCSA, ETSI, TTA and TTC who jointly own the copyright in them

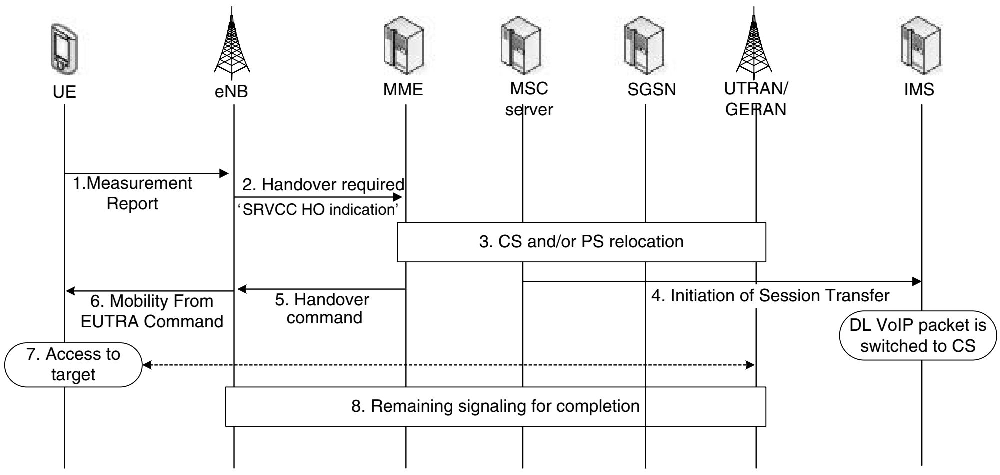

Figure 8.7 SRVCC handover from E-UTRAN to UTRAN/GERAN. Reproduced by permission of 3GPP, © 2011. 3GPP^{TM} TSs and TRs are the property of ARIB, ATIS, CCSA, ETSI, TTA and TTC who jointly own the copyright in them

- **Step 1:** The UE sends a *MeasurementReport* message including measured results of other RAT to its serving cell.
- **Step 2:** The eNB, receiving the *MeasurementReport* message, decides whether SRVCC should be triggered based on whether any voice bearer, for which QCI is set to 1, is established. If SRVCC is triggered, the eNB requests an SRVCC handover to the MME by sending Handover Required including an SRVCC handover indication.
- **Step 3:** The MME, receiving the SRVCC handover indication, splits the voice bearer from other non-voice bearers. The MME then initiates relocation of the voice bearer and other non-voice bearers towards the MSC server and SGSN, respectively. The details of relocation can be found in [3]. During this step, radio resources for the bearers are assigned for the UE by the RNC of the target cell. Upon completion of relocation, the MME receives acknowledgments for the relocations from the MSC server and SGSN.
- **Step 4:** The MSC server, upon receiving the acknowledgment for the relocation request for the voice bearer from the target MSC in step 3, initiates a session transfer towards the IMS. This step is carried out during step 3.
- **Step 5:** The MME, upon receiving acknowledgments both for the relocation of the voice bearer and the non-voice bearers, sends a handover command to the eNB. The handover command includes the information (Target to Source Transparent Container) received from the SGSN and MSC server during step 3.
- **Step 6:** The eNB, upon receiving the handover command, sends to the UE a *MobilityFromEUTRACommand* message.
- **Step 7:** The UE, upon receiving the *MobilityFromEUTRACommand*, accesses the target cell.
- **Step 8:** Further signaling for completion of the SRVCC handover is exchanged, for example, further confirmations between the nodes, release of resources, and so on. Details can be found in [3].

8.3 Circuit-Switched Fallback (CSFB)

Circuit-Switched FallBack (CSFB) is the mechanism, defined in Release 8, to enable a voice service by reusing an existing CS network for UEs normally camping on LTE for IP services. If CSFB is used when a voice service is to be initiated, the UE performs radio switching from LTE to a legacy 2G/3G CS network where a CS voice call can be established. CSFB is applicable to UEs supporting both the LTE network and CS networks such as GERAN, UTRAN, or CDMA2000 1xRTT. From a network point of view, the coverage of a legacy CS network is much greater than the LTE coverage, such that a UE can be directed to one of the available legacy CS networks when CSFB is initiated. The general architectural enhancement to enable CSFB and relevant functionalities/procedures is specified in [4].

CSFB is initiated by a UE sending a particular NAS message, called an extended service request, to the MME. The extended service request may be sent in response to a paging message that was originated in the CS domain and routed to E-UTRAN via the MME in the case of a mobile terminating call. Upon receiving the extended service request for CSFB, the MME requests the eNB to trigger an inter-RAT mobility procedure. The details of CSFB procedures are explained in the subsequent sections.

The inter-RAT mobility procedures applicable for CSFB include redirection, handover, and cell change order:

- **Redirection:** The redirection procedure can be performed by the eNB sending an *RRCConnectionRelease* message including redirection information (see Section 3.12). The redirection information indicates the RAT/frequency to which the UE needs to move. Upon receiving redirection information, the UE releases the RRC connection with E-UTRAN, performs cell selection to camp on a cell on the indicated RAT/frequency, and makes a connection with the selected cell. The redirection procedure is applicable for CSFB to GERAN, UTRAN, and CDMA2000 1xRTT.
- **Handover:** Handover to the other RAT is performed by the eNB issuing a *MobilityFromEUTRACommand* message as a handover command. The handover command includes the radio resource configuration to be used in the target cell. If a UE receives a handover command, it attempts to access the target cell using the radio resource configuration included in the handover command. Handover to the other RAT is applicable for CSFB to GERAN, UTRAN, and CDMA2000 1xRTT. If the *MobilityFromEUTRACommand* message is sent for CSFB to GERAN or UTRAN, it includes a CSFB indicator for use in NAS to handle a handover failure.
- **Cell Change Order:** Cell Change Order (CCO) can also be used for CSFB. CCO is applicable only for CSFB to GERAN. CCO is signaled to the UE via a *MobilityFromEUTRACommand* including the information relevant for CCO.

The mobility options applicable for CSFB to UTRAN/GERAN are summarized in Table 8.1.

Note that the RRC connection in E-UTRAN is not maintained after CSFB. This means that, while the UE is provided with a voice service in the legacy RAT, IP services may be degraded or even suspended, depending on the support for IP services in the legacy RAT. To provide a better quality of IP services to the UE, the legacy RAT may decide to perform inter-RAT mobility to E-UTRAN for the UE as soon as possible

Table 8.1 Possible mobility options for CSFB to UTRAN/GERAN. Reproduced by permission of 3GPP, © 2011. 3GPP™ TSs and TRs are the property of ARIB, ATIS, CCSA, ETSI, TTA and TTC who jointly own the copyright in them

No	Mobility Option (Related RRC Messages)	Target RAT	Release	UE Capability
1	Redirection (*RRCConnectionRelease*)	UTRAN	Release 8	Mandatory for UEs supporting CSFB to the UTRAN
2	Redirection with system information (*RRCConnectionRelease*)	UTRAN	Release 9	Indication of the *e-RedirectionUTRA* in the UE capability
3	Redirection (*RRCConnectionRelease*)	GERAN	Release 8	Mandatory for UEs supporting CSFB to the GERAN
4	Redirection with system information (*RRCConnectionRelease*)	GERAN	Release 9	Mandatory for UEs supporting CSFB to the GERAN
5	PS handover with DRB(s) (*MobilityFromEUTRACommand*)	UTRAN	Release 8	Mandatory for UEs supporting CSFB to the UTRAN
6	PS handover (*MobilityFromEUTRACommand*)	GERAN	Release 8	Indication of the *interRAT-PS-HO-ToGERAN* in the UE capability
7	Cell change order with the NACC (*MobilityFromEUTRACommand*)	GERAN	Release 8	Mandatory for UEs supporting CSFB to the GERAN
8	Cell change order without the NACC (*MobilityFromEUTRACommand*)	GERAN	Release 8	Mandatory for UEs supporting CSFB to the GERAN

after the end of the voice service. No UE autonomous action for reverting to E-UTRAN is defined.

Compared to the setup time for a voice call within a legacy CS network from scratch, CSFB introduces more delay to establish a voice call, mainly due to radio switching. In Release 9, some features were added to enhance CSFB performance.

A UE that is capable of CSFB to UTRAN indicates support for UTRA FDD or TDD and the supported band list in the UE capability signaling. A UE capable of CSFB to GERAN indicates support for GERAN and the supported band list in the UE capability signaling.

8.3.1 CSFB to UTRAN or GERAN

8.3.1.1 Reference Architecture

Figure 8.8 shows the network architecture for CSFB to UTRAN/GERAN. The MME and MSC server are interconnected via the SGs interface, over which signaling exchange and the delivery of CS paging (paging that originated from a CS network) to the LTE network are performed.

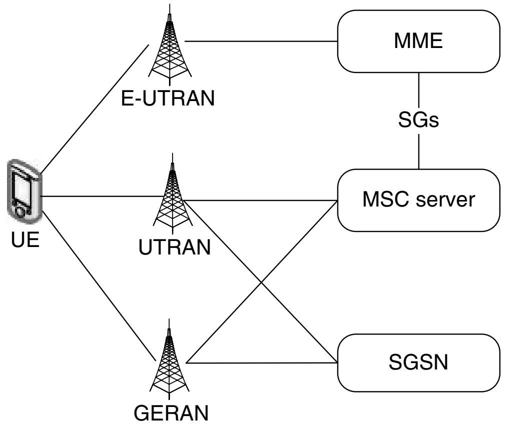

Figure 8.8 Network architecture for CSFB to UTRAN/GERAN

8.3.1.2 Combined Registration to the LTE Network and CS for CSFB

In CSFB, since a voice service is delivered over 2G/3G CS networks, the UE should first be registered to both the LTE network and the 2G/3G CS network to enable CSFB. Registration to both the LTE network and the CS network is done efficiently with a "combined" registration procedure, rather than performing two registration procedures independently. The combined registration can be applied to the Attach procedure and the Location Area Update procedure in the NAS layer.

With regard to the combined registration, the mapping between tracking area and location/routing area is managed in the MME, which enables the MME to trigger CSFB towards the proper cell of the CS network.

8.3.1.3 CSFB for a Mobile Originating Call

Figure 8.9 shows a simplified illustration of a mobile originating call with CSFB procedures. Each step of the procedure is described below:

- **Step 1:** If the UE cannot use IMS VoIP in LTE and the UE is attached to both LTE and CS networks, it triggers CSFB for a mobile originating call by sending an extended service request to the MME.
- **Step 2:** The MME, receiving the extended service request, triggers the eNB to perform the CSFB-related inter-RAT mobility procedure from E-UTRAN to GERAN or UTRAN by using an S1AP message. In this S1AP message, a CSFB indicator and the Location Area Identity (LAI) to which the UE is registered are included.
- **Step 3:** The eNB, receiving the message triggering CSFB, triggers an inter-RAT mobility procedure to GERAN or UTRAN. There are many options for carrying out inter-RAT mobility: PS handover, redirection, and cell change order. Depending on the selected mobility procedure, the eNB sends a relevant command for inter-RAT mobility. A *MobilityFromEUTRACommand* message is sent to the UE for handover or cell change order,

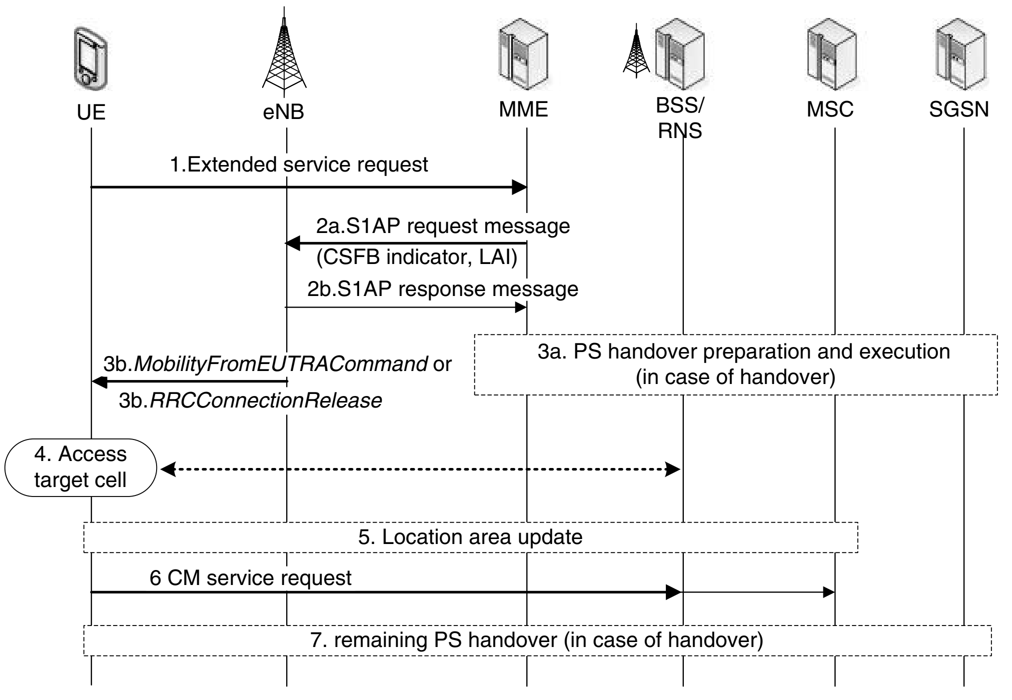

Figure 8.9 Mobile originating call with CSFB to UTRAN/GERAN. Reproduced by permission of 3GPP, © 2011. 3GPP™ TSs and TRs are the property of ARIB, ATIS, CCSA, ETSI, TTA and TTC who jointly own the copyright in them

and an *RRCConnectionRelease* message including redirection information is sent for redirection.

- **Step 4:** The UE, receiving the inter-RAT mobility command from the eNB, performs an inter-RAT mobility procedure. If the UE receives a command for handover, it synchronizes and accesses the target using the radio resource configuration included in the handover command. If the UE receives a command for cell change order or redirection, the UE selects a suitable cell on the RAT/frequency indicated in the command and attempts to access the cell.
- **Step 5:** If the new cell belongs to a location area that is different from the cell the UE has most recently registered to, the UE updates its location to the CS network.
- **Step 6:** If the inter-RAT mobility procedure is successful, the UE continues the CS voice call setup procedure by sending a Connection Management (CM) service request. In Release 10, the CM service request indicates that this is a call establishment as a result of CSFB.
- **Step 7:** In cases where PS handover is needed, the remaining steps to complete the handover are processed.

Figure 8.10 shows the change of routing path for data and voice due to a mobile originating CSFB call.

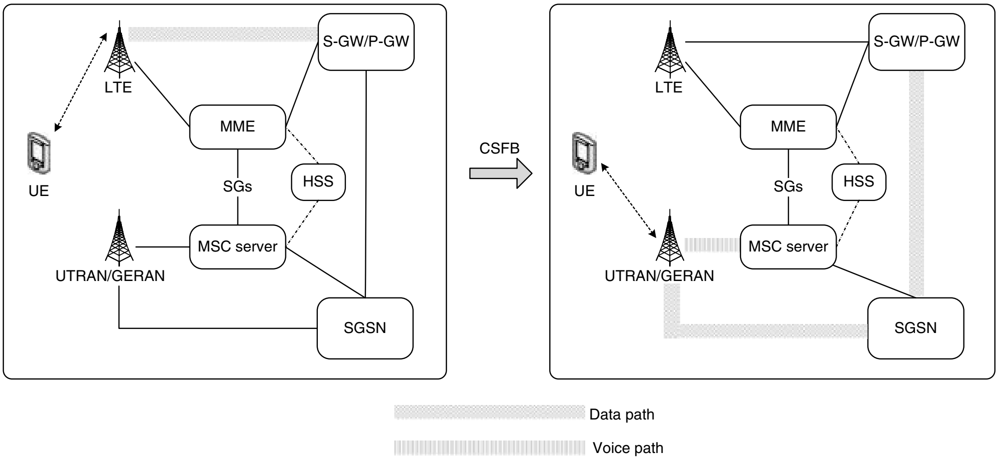

Figure 8.10 Change of voice and data path after CSFB

8.3.1.4 CSFB for a Mobile Terminating Call

The procedures for a mobile terminating call with CSFB to UTRAN/GERAN are shown in Figure 8.11. In the mobile terminating call, a paging procedure is added to the mobile terminating call. A mobile terminating call involving CSFB is initiated by the MSC server receiving an incoming voice call.

- **Step 1:** If the MSC receives an incoming voice call, it sends a paging request to the MME via the SGs interface. The MME, upon receiving the paging request from the MSC, pages the UE unless the UE is configured only for SMS. Note that the *Paging* message includes a CN domain indicator set to "CS".
- **Step 2:** The UE, receiving the *Paging* message addressed to it, establishes an RRC connection if in RRC_IDLE and sends an extended service request for CSFB to the MME. The MME, receiving the extended service request, sends a service request over the SGs to the MSC. If the MSC receives a service request from the MME via the SGs, it stops retransmitting the paging request to the MME.
- **Step 3:** The MME then triggers the eNB to perform an inter-RAT mobility procedure for the UE by sending an S1AP message including a CSFB Indicator and LAI allocated to the UE. The S1AP message further includes UE capabilities to assist the eNB mobility procedure.
- **Steps 4–8:** For the rest of the process, the UE follows almost the same procedures as those shown in Figure 8.9, except for step 7, where the UE sends a paging response instead of a CM service request. If the MSC receives a paging response, it establishes a voice call.

8.3.1.5 Performance Enhancement of CSFB to GERAN/UTRAN

In Release 8, the basic mechanisms of CSFB were defined. Compared to the call setup time of a native voice call that originates and then terminates purely in the CS domain without radio switching, the CSFB mechanism introduces some additional delay when establishing a voice call. There are several causes contributing to such additional delay during CSFB:

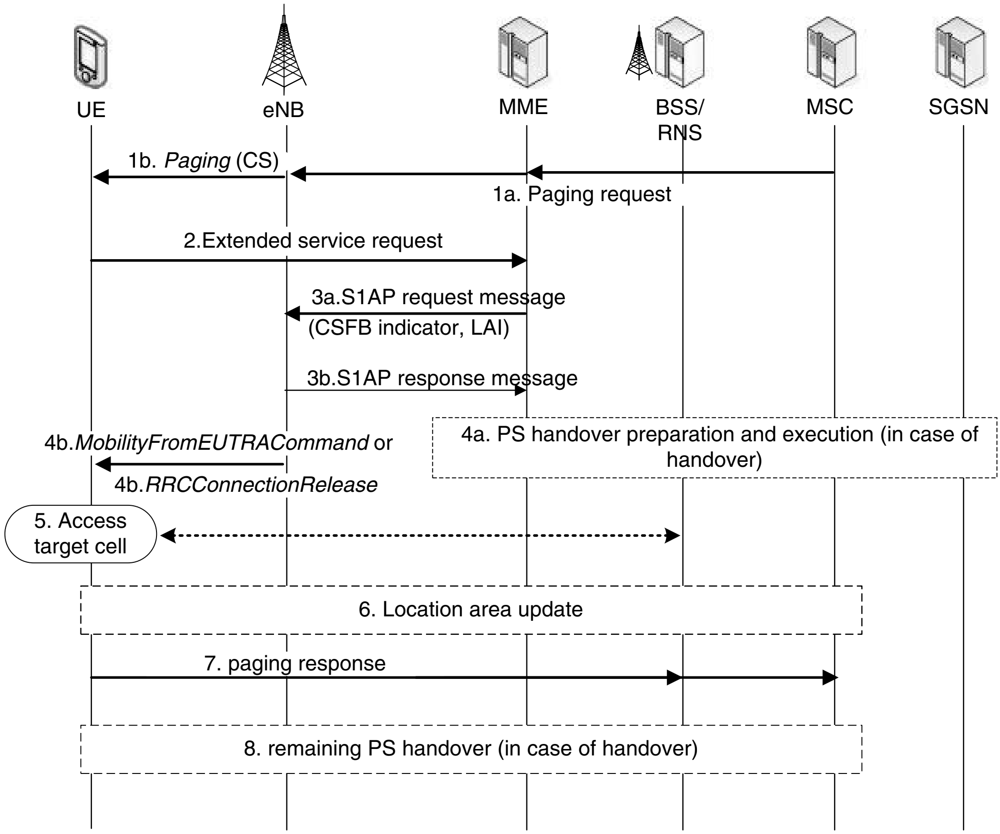

Figure 8.11 Mobile terminating call with CSFB to UTRAN/GERAN. Reproduced by permission of 3GPP, © 2011. 3GPP™ TSs and TRs are the property of ARIB, ATIS, CCSA, ETSI, TTA and TTC who jointly own the copyright in them

- **Inter-RAT measurements:** If PS handover is used for CSFB, the UE may be requested by the eNB to perform inter-RAT measurements and reporting in order to select a suitable target cell. In this case, the inter-RAT measurement procedure introduces non-trivial delay.
- **System information acquisition of inter-RAT cell:** If redirection is used for CSFB, the UE needs to search for a suitable cell on inter-RAT. From the selected cell of the inter-RAT, the UE has to acquire the radio resource configuration parameters and the accessibility parameters from the system information. It takes noticeable time to read the system information of the selected cell, for example, $1 \sim 2$ seconds to acquire the relevant system information of a UTRAN cell, depending on the scheduling of the relevant system information on the radio interface.
- **Location area update at inter-RAT:** After executing inter-RAT mobility for CSFB, the NAS layer of the UE needs to trigger a Location Area Update procedure if the new cell belongs to an LAI that is different from that stored in the UE.

To enhance the performance of the CSFB procedure, Release 9 introduced an optimized redirection mechanism whereby an eNB can include the system information of one or multiple target cells (up to six) in the *RRCConnectionRelease* message. The system information can be utilized by the UE to accelerate the redirection procedure during CSFB, by allowing the UE to skip system information acquisition on the selected cell.

With regard to the possible delay incurred by inter-RAT measurements, the network may perform handover blindly (i.e., without a measurement report from the UE) to reduce the delay. However, blind handover may decrease the rate of handover success, which in turn increases the call drop rate.

The occurrence of a location area update during CSFB can be minimized if the network carefully performs coverage engineering jointly with the tracking area and the location/routing area. One example of coverage engineering is where the cell coverage of E-UTRAN/UTRAN/GERAN and the corresponding tracking/location/routing area are fine-tuned in such a way that whenever the location/routing area is changed, the tracking area is also changed.

8.3.2 CSFB to CDMA2000 1xRTT

3GPP also accommodates CDMA2000 1xRTT CS defined by 3GPP2 as one of the target CS domains for CSFB, and hence both mobile originating calls and mobile terminating calls via CSFB to CDMA2000 1xRTT CS are supported from Release 8.

8.3.2.1 Mechanisms of CSFB to 1xRTT

There are several mechanisms of CSFB to CDMA2000 1xRTT, which include (1) 1xCSFB, (2) enhanced 1xCSFB (e1xCSFB), (3) 1xCSFB with dual receiver, and (4) e1xCSFB with dual transceiver, depending on inter-RAT mobility type and UE capability. For CSFB to CDMA 1xRTT, the eNB provides the 1xRTT-related parameters in SIB8 of the system information.

1. **1xCSFB:** 1xCSFB is the default mechanism defined in Release 8 for CSFB to 1xRTT. Redirection is used for inter-RAT mobility triggered by 1xCSFB; that is, the eNB sends an *RRCConnectionRelease* message with redirection information to the UE for inter-RAT mobility from E-UTRAN to 1xRTT. Upon receiving the mobility command for 1xCSFB, the UE releases the RRC connection, leaves E-UTRAN, and attempts to access the 1xRTT. If 1xCSFB is performed, IP services over EPS bearers are suspended or deactivated.

   The network support for 1xCSFB is indicated by providing 1xRTT registration parameters in *SystemInformationBlock8* (SIB8) of an E-UTRAN cell. Support for 1xCSFB is mandatory for a UE supporting CSFB to 1xRTT. A 1xCSFB-capable UE registers with 1xRTT via the LTE network by using the 1xRTT registration parameters provided in SIB8.
2. **Enhanced 1xCSFB:** Enhanced 1xCSFB (e1xCSFB) is the enhanced version of 1xCSFB defined in Release 9 to enable the network to provide the UE with radio resources that are allocated for the UE by the target 1xRTT cell.

The UE indicates to the network support for e1xCSFB by *e-CSFB-1XRTT* in the UE capability. Based on this capability information, the eNB may decide to perform e1xCSFB for the UE when a voice service is to be initiated. A UE supporting e1xCSFB registers with 1xRTT according to the 1xRTT registration parameters in SIB8.

e1xCSFB allows the option of performing concurrent PS handover to High Rate Packet Data (HRPD), to provide IP service continuity after e1xCSFB. For enhanced 1xCSFB with concurrent PS handover to HRPD, the UE indicates that it supports concurrent 1xRTT and HRPD to the network in the UE capability.

For e1xCSFB, the eNB sends to the UE a *MobilityFromEUTRACommand* message that includes 1xRTT configuration; that is, a 1xRTT channel assignment message. If concurrent PS handover to HRPD needs to be performed, the *MobilityFromEUTRACommand* also includes HRPD redirection information. Upon receiving the mobility command, the UE attempts to access the 1xRTT cell by using the 1xRTT channel assignment information.

3. **1xCSFB with Dual Receiver:** The dual receiver 1xCSFB defined in Release 9 is the approach that makes use of dual radio capabilities of the UE, if supported. The dual receiver capability makes it possible for the UE to receive over E-UTRAN and 1xRTT simultaneously. In this case, the UE normally transmits and receives on E-UTRAN while the UE is also camping on 1xRTT performing 1xRTT dormant mode operations by using 1xRTT receiving capability.

   Note that registration to 1xRTT is done directly over 1xRTT without signaling through the LTE network. This means that simpler network implementation is possible in 1xCSFB with dual receiver because coordination between the 1xRTT and the LTE network for 1xRTT registration is not necessary. When the UE needs to leave E-UTRAN for registration to 1xRTT, the UE sends an extended service request to the MME, similar to the normal CSFB MT/MO call procedures, which results in the release of the RRC connection to the eNB and an S1-U bearer to the S-GW.

   Mobility to 1xRTT is performed by the RRC connection release procedure. The *RRCConnectionRelease* message does not include redirection information because the UE has already been camping on 1xRTT after registration to 1xRTT by using the CDMA2000 radio receiver. It should be noted that with this dual receiver 1xCSFB, UE battery consumption will be higher because both E-UTRAN and 1xRTT radios should be kept running.
4. **e1xCSFB with Dual Transceiver:** If the UE is capable of receiving/transmitting over both LTE and 1xRTT radios simultaneously, the simplest way of providing CS voice services would be to make both radios operate independently. The benefit of independent operations is that the IP service over LTE is not impacted by the status of 1xRTT radio operations. However, such approach would considerably shorten UE battery life as two radios need to be running all the time.

   In Release 10, a dual transceiver e1xCSFB mechanism is introduced to utilize the dual transceiver capability whilst avoiding higher battery consumption. The idea behind the dual transceiver e1xCSFB mechanism is to turn on the 1xRTT radio only when a CS voice call or SMS are to be served, and to turn it off otherwise. While the 1xRTT radio is switched on, the 1xRTT signaling for, for example, registration to 1xRTT is tunneled between the UE and 1xRTT via the LTE network. Note that even while the 1xRTT radio

Table 8.2 Possible mobility options for 1xCSFB. Reproduced by permission of 3GPP, © 2011. 3GPP™ TSs and TRs are the property of ARIB, ATIS, CCSA, ETSI, TTA and TTC who jointly own the copyright in them

No	Mobility Option (Related RRC Messages)	Tx/Rx	Release	UE Capability
1	Redirection (*RRCConnectionRelease* with redirection information)	Single/Single	Release 8	Mandatory for UEs supporting CSFB to 1xRTT
2	Enhanced 1xCSFB (*HandoverFromEUTRA PreparationRequest, ULHandoverPreparationTransfer, MobilityFromEUTRACommand*)	Single/Single	Release 9	Indication of the *e-CSFB-1XRTT* in the UE capability
3	Enhanced 1xCSFB with concurrent HRPD handover (same as enhanced 1xCSFB)	Single/Single	Release 9	Indication of the *e-CSFB-ConcPS-Mob1XRTT* in the UE capability
4	1xCSFB with dual receiver (*RRCConnectionRelease* without redirection information)	Single/Dual	Release 9	Indication of the *rx-Config1XRTT* set to *dual* in the UE capability
5	Enhanced 1xCSFB with dual transceiver (*HandoverFromEUTRA PreparationRequest, ULHandoverPreparationTransfer, DLInformationTransfer*)	Dual/Dual	Release 10	Indication of the *e-CSFB-dual-1XRTT* in the UE capability

is running, the UE can keep connected with E-UTRAN to continue IP services with dual transceiver capability.

The eNB advertises support for e1xCSFB with dual transceiver by broadcasting *csfb-DualRxTxSupport* in SIB8. The UE indicates support for e1xCSFB with dual transceiver by *e-CSFB-dual-1XRTT* in the UE capability. For e1xCSFB with dual transceiver, the eNB may send to the UE a *DLInformationTransfer* message including the 1xRTT configuration. Upon receiving the *DLInformationTransfer* message, the UE with dual transceiver turns on the 1xRTT radio while the UE maintains the operations of the LTE radio.

Table 8.2 summarizes the available options for 1xCSFB.

8.3.2.2 Reference Architecture

Figure 8.12 shows the network architecture for CSFB to 1xRTT. The MME is connected to the 1xCS Interworking System (IWS) via the S102 interface, over which tunneled 1xRTT messages are communicated between the 1xRTT part of the UE and the 1xRTT network. 1xRTT CS paging is also transferred to the LTE network via S102 and is then delivered to the UE.

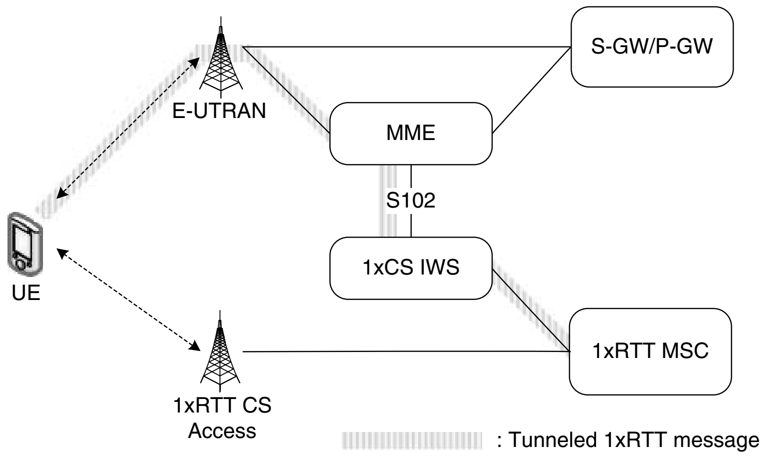

Figure 8.12 Network architecture for CSFB to 1xRTT CS. Reproduced by permission of 3GPP, © 2011. 3GPP™ TSs and TRs are the property of ARIB, ATIS, CCSA, ETSI, TTA and TTC who jointly own the copyright in them

8.3.2.3 Pre-Registration to 1xRTT over the LTE Network

A UE needs to be registered on a 1xRTT network for CSFB to 1xRTT. In Release 8, a mechanism was introduced to allow a UE to register with 1xRTT via the LTE network. This 1xRTT registration procedure over the LTE network is often called "pre-registration" and is performed by exchanging 1xCS messages between the UE and the 1xRTT MSC. Pre-registration can be applied for 1xRTT CS and/or 1xRTT HRPD.

Figure 8.13 illustrates the pre-registration procedure. The E-UTRAN can advertise the pre-registration parameters in SIB8. Network support for pre-registration to 1xRTT and HRPD is indicated by *csfb-RegistrationParam1xRTT* and *preRegistrationInfoHRPD*, respectively, in SIB8.

The RRC layer of the UE, upon receiving a SIB8 indicating support for pre-registration, forwards the relevant parameters included in SIB8 to the CDMA2000 upper layer to determine whether (re-)registration is needed. If the CDMA2000 upper layer of the UE requests registration to 1xRTT CS, the UE performs the registration procedure by sending a 1xRTT CS registration request message to the eNB. The 1xRTT CS registration request message is forwarded to the MME and then tunneled via the S102 interface to the 1xCS IWS responsible for interworking between the LTE network and the 1xRTT CS domain.

Upon receiving the 1xRTT CS registration request message, the 1xCS IWS performs registration of the UE with the 1xRTT MSC. The result of 1xRTT CS registration is sent back to the UE through a 1xRTT CS registration response message.

8.3.2.4 1xRTT CSFB for a Mobile Originating Call

Figure 8.14 shows the simplified procedures of a mobile originating call with CSFB to 1xRTT using the *RRCConnectionRelease* message with redirection information.

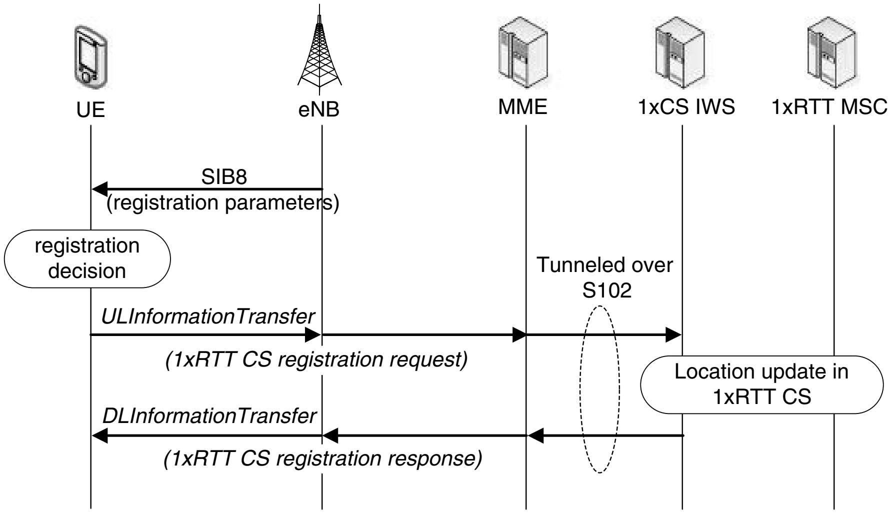

Figure 8.13 Pre-registration to 1xRTT. Reproduced by permission of 3GPP, © 2011. 3GPP™ TSs and TRs are the property of ARIB, ATIS, CCSA, ETSI, TTA and TTC who jointly own the copyright in them

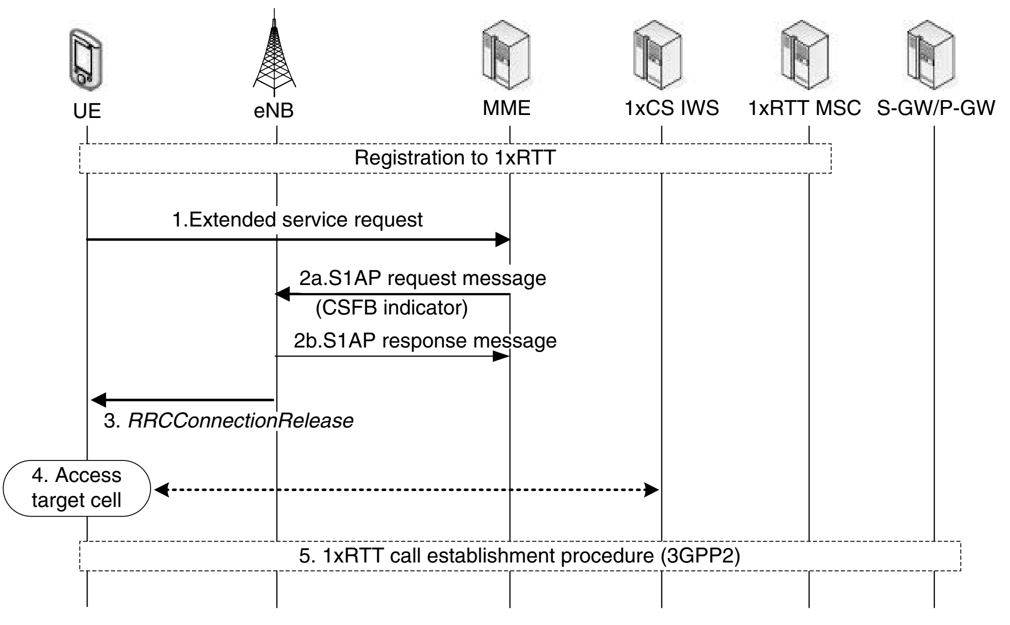

Figure 8.14 Mobile originating call with CSFB to 1xRTT (1xCSFB). Reproduced by permission of 3GPP, © 2011. 3GPP™ TSs and TRs are the property of ARIB, ATIS, CCSA, ETSI, TTA and TTC who jointly own the copyright in them

- **Step 1:** If the UE cannot initiate IMS VoIP and the UE is attached to both LTE and 1xRTT, the UE decides to invoke 1xRTT CSFB for a mobile originating call by sending an extended service request to the MME.
- **Step 2:** The MME, receiving the extended service request, triggers the eNB to perform the CSFB-related inter-RAT mobility procedure from E-UTRAN to 1xRTT by using an S1AP message (UE context modification request). The S1AP message includes a CSFB indicator.
- **Step 3:** The eNB, upon receiving the S1AP message including the CSFB indicator, triggers inter-RAT mobility to 1xRTT by sending an *RRCConnectionRelease* message to the UE including redirection information towards 1xRTT. Then, the eNB requests that the MME release the S1 UE context stored in the MME.
- **Step 4:** The UE, upon receiving the *RRCConnectionRelease* message from the eNB, releases the RRC connection and accesses the 1xRTT network according to the redirection information included in the *RRCConnectionRelease*.
- **Step 5:** For 1xRTT call establishment, the UE follows 3GPP2 specifications.

8.3.2.5 Enhanced 1xRTT CSFB (e1xCSFB) for a Mobile Originating Call

Figure 8.15 shows the simplified procedures for e1xCSFB for a mobile originating call. Note that in step 7, a *MobilityFromEUTRACommand* is sent to a UE supporting a single transceiver while a *DLInformationTransfer* is sent to a UE supporting a dual transceiver.

8.3.2.6 1xRTT CSFB for a Mobile Terminating Call

For a mobile terminating call with CSFB to 1xRTT, paging reception/response signaling is added to the procedure for a mobile terminating call. Figure 8.16 shows the 1xRTT CSFB procedures for a mobile terminating call.

- **Step 1:** The 1xRTT MSC sends a 1xRTT CS paging request to the 1xCS IWS. The paging request is then tunneled to the MME via the S102 interface. The 1xRTT CS paging request is forwarded to the eNB and then forwarded to the UE via a *DLInformationTransfer*.
- **Step 2:** The UE responds to the received 1xRTT CS paging request by sending an extended service request to the MME.
- **Step 3:** The MME, receiving the extended service request, invokes the eNB to perform CSFB-related inter-RAT mobility by using an S1AP message including a CSFB indicator.
- **Steps 4–7:** The same steps 3 to 5 in Figure 8.14 are applied. In addition, the UE sends a 1xRTT CS paging response to 1xRTT. Upon receiving the paging response, the 1xRTT MSC stops retransmitting the 1xRTT CS paging request.

8.4 Service Domain Selection

For a mobile originating call, if the UE supports both CSFB and IMS VoIP, it has to decide which voice solution should be used out of preference. This decision is equivalent to selecting either the CS or PS domain for voice services. The UE desirably considers (1) the voice domain preference provisioned to the UE by the HPLMN operator's preference, and (2) the network capabilities indicated by the network [5].

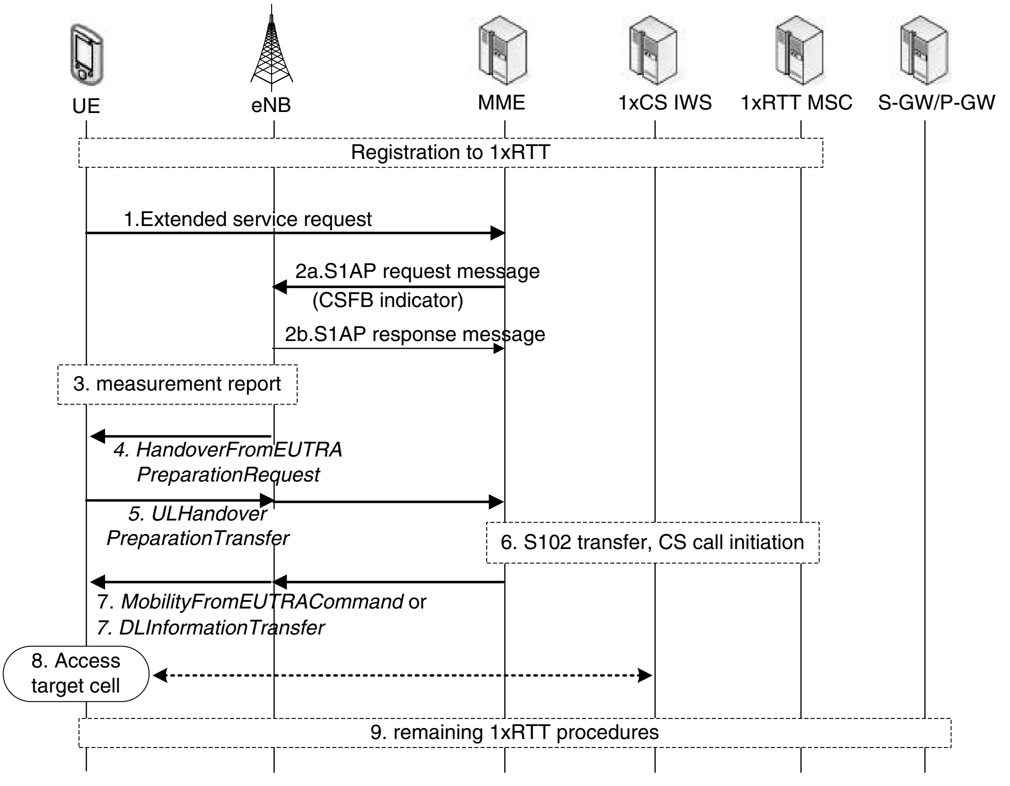

Figure 8.15 Mobile originating call with enhanced CSFB to 1xRTT. Reproduced by permission of 3GPP, © 2011. 3GPP™ TSs and TRs are the property of ARIB, ATIS, CCSA, ETSI, TTA and TTC who jointly own the copyright in them

- **Voice Domain Preference:** The voice domain preference indicates the domain (PS domain or CS domain) preferred by the UE for a voice service.
  - **CS Voice only:** The UE configured with this value does not attempt to use IMS voice with an EPS bearer.
  - **CS Voice preferred, IMS PS Voice as secondary:** The UE configured with this value "preferably" attempts to use the CS domain for a voice service.
  - **IMS PS Voice preferred, CS Voice as secondary:** The UE configured with this value "preferably" attempts to use IMS voice with an EPS bearer.
  - **IMS PS Voice only:** The UE configured with this value attempts to use IMS voice with an EPS bearer.

  The UE provides this information to the network, and the network can use the information to select a RAT/Frequency Selection Priority (RFSP) for the UE. The selected RFSP may be configured to the UE in order to steer the UE towards the RAT of the preferred domain.

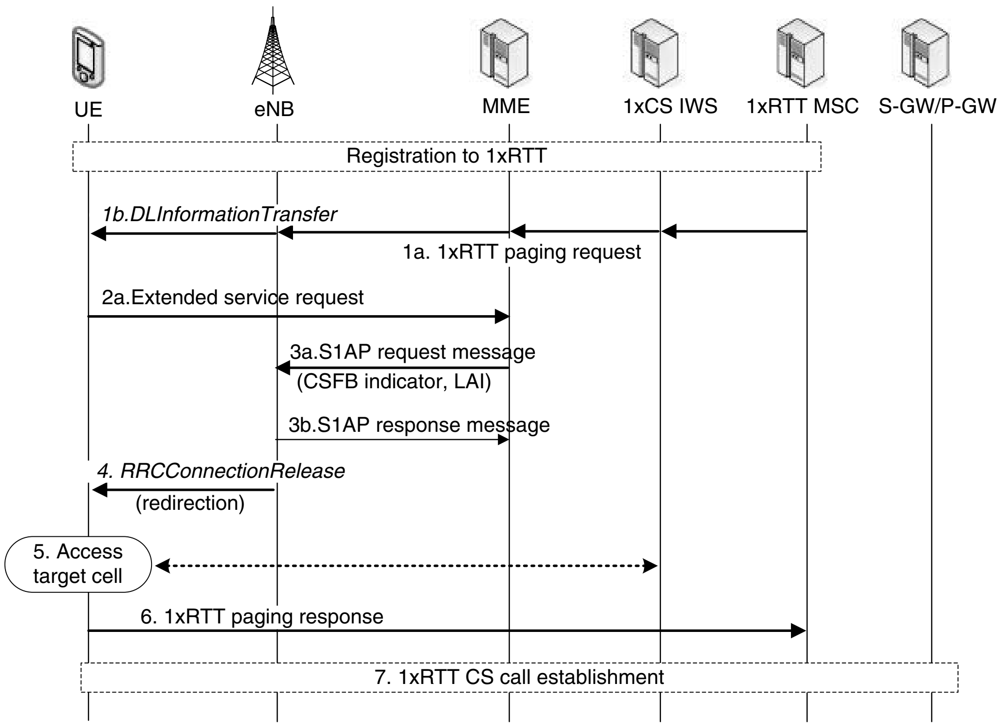

Figure 8.16 Mobile terminating call with CSFB to 1xRTT (1xCSFB). Reproduced by permission of 3GPP, © 2011. 3GPP™ TSs and TRs are the property of ARIB, ATIS, CCSA, ETSI, TTA and TTC who jointly own the copyright in them

- **Network Capabilities:** During the Attach or Tracking Area Update procedures, the network provides the UE with the following information on its supported features. This information may affect the selection of voice solution by the UE:
  – IMS voice support;
  – SMS only;
  – CSFB not preferred.

8.4.1 UE Decision between IMS VoIP and CSFB

In principle, the choice between IMS voice and CSFB is left to UE implementations, but the implementation guidelines for UEs supporting both CSFB and IMS are presented in [6]. In general, the UE knows, during the Attach procedure(s), whether its preferred domain for a voice service can actually provide a voice service. If a voice service is possible on the preferred domain, the UE will use the preferred domain when a voice call is to be serviced, otherwise it will use another domain. For example, let us assume that one UE is set to "CS Voice preferred, IMS PS Voice as secondary". According to the UE's voice domain preference, the UE first attempts to initiate combined registration to be ready for CSFB, as described in Section 8.3. If the combined attach procedure is successful, and if the network does not indicate “CSFB not preferred” or “SMS only” in the attach accept message, the UE will desirably select CSFB for a voice service. Otherwise, the UE will use IMS voice. Other examples can be found in [6].

8.5 Comparison between IMS VoIP and CSFB

IMS VoIP can fully employ the benefits of a flat IP architecture, for example, maintaining the LTE data rate during a voice call and accommodating fruitful multimedia attributes that can be blended with a voice service or other services. Investment on the network is focused largely on deploying the IMS core rather than upgrading a legacy network. Since the voice service is integrated with the generic IMS framework within all IP structures, operational benefit from simplified service provision is maximized. IMS VoIP is fully 3GPP-compliant, and thus fewer interoperability issues are expected.

CSFB induces additional call setup delay in comparison to a generic voice call that originates and terminates in a legacy network. The contributors to the additional delay include the RRC Connection Establishment procedure on LTE and the subsequent mobility procedure from LTE to 2G/3G, which are not required in a legacy voice call setup procedure. If the UE needs to perform a location area update after moving to legacy RAT during CSFB, further delay is added. During a voice call enabled by CSFB, it is unavoidable that the data service will be degraded or even suspended as the UE is served by legacy RAT. CSFB also requires an upgrade of a legacy network to support, for example, CS paging over E-UTRAN and mobility from LTE to 2G/3G for CSFB.

Even with its shortcomings, CSFB is a viable and appealing solution because it can readily provide voice services by reusing the existing voice solution over the legacy network. The changes required to the network are not significant, so operators are willing to promote the use of CSFB as an interim voice solution. Since CSFB is a 3GPP standard, interoperability issues can be mostly avoided.

In Table 8.3, a summary of comparison results between IMS VoIP and CSFB is provided.

8.6 RAN Optimization for VoIP

From an access stratum perspective, the bearer carrying voice packets has the following characteristics:

- the payload size of voice packets is small;
- the arrival of IP packets is regular, for example, with 20 ms periodicity.

From these characteristics, one would expect the signaling overhead due to a relatively larger packet header and frequent scheduling to result in a significant increase of signaling overhead in the radio interface when VoIP is used extensively. In some situations, VoIP coverage may be limited by a power-limited uplink, which cannot be resolved by conventional retransmission mechanisms due to the stringent delay requirement for voice. To resolve these issues, Release 8 defined some optimizations including Robust Header Compression, TTI bundling, and Semi-Persistent Scheduling.

Table 8.3 Comparison between IMS VoIP and CSFB

Comparison Class	Comparison Factor	IMS VoIP	CSFB
Performance	Call setup delay	Comparable or shorter than native call setup delay	Longer than native call setup delay
	Concurrent voice and data during voice call	Yes	Yes/No (depending on target)
	Maintaining LTE data rate during voice service	Yes	No
Service	Support of advanced services	Yes	No
UE complexity	Features to be supported	Medium complexity (IMS client, SRVCC)	Low complexity (CSFB-related enhancement)
NW operational benefit/cost	New network equipment installation	Yes (IMS core)	None, but small upgrades to MME
	Legacy network equipment upgrade	Limited	Yes (all MSCs overlaid on LTE networks)
	Coverage engineering between LTE and legacy coverage	Not required	Required
	All IP structure	Yes	No
Interoperability issue	Standard compliance	3GPP compliant	3GPP compliant

8.6.1 Robust Header Compression (ROHC)

ROHC is a method to compress IP/UDP/RTP or IP/TCP headers. The IMS Profile for Voice and SMS mandates that the UE and the network should support Robust Header Compression (ROHC) to reduce overhead. ROHC is performed in the PDCP layer. The details of ROHC are described in Section 4.2.

8.6.2 TTI Bundling

In cases where the coverage of VoIP is limited by a power-limited uplink at, for example, a cell edge, conventional methods such as HARQ retransmission in the MAC layer or segmentation in the RLC layer could be used. Such methods cannot avoid introducing additional delay and signaling overhead, which definitely impacts negatively upon the quality of real-time service and efficient usage of scarce radio resources.

As an alternative solution, the concept of TTI bundling has been introduced. If TTI bundling is configured, one transport block containing a VoIP packet is transmitted repeatedly over a bundle of four consecutive TTIs. In each TTI, a different redundancy version is applied to the transport block. Downlink HARQ feedback for uplink transmission with TTI bundling is sent by the eNB once per bundle. For more information, see Section 6.8.

8.6.3 Semi-Persistent Scheduling for HARQ

Since a speech codec typically generates a voice packet every 20 ms, a scheduling grant needs to be provided to the UE in a periodic manner during a voice service. Considering that the size of a VoIP packet is relatively small, dynamic scheduling of each voice packet would result in significant overhead in terms of control signaling and scheduling.

To reduce such overhead, Semi-Persistent Scheduling was introduced in Release 8. With Semi-Persistent Scheduling, predefined scheduling grants for reception and/or transmission are configured, and the UE can receive downlink traffic or transmit uplink traffic at predefined occasions without explicit signaling. For more information, see Section 6.4.2.

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