

GERAN – E-UTRAN Interworking Basics & Status

TSG GERAN#34

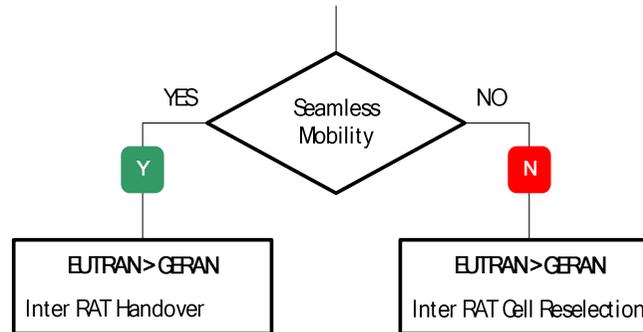
Shenzen, China

14th -18th of May 2007

Source: Nokia Siemens Networks / Nokia

Interworking Mechanisms

GERAN / LTE Interworking mechanisms for supporting *Mobility* and *Seamless Mobility*



- Cell Reselection procedures provide *mobility*
 - Performance good enough for non-real time, interactive, best effort service continuity
 - GERAN/LTE Inter-RAT cell reselection procedures will be specified as done for GERAN/UTRAN
- Handover procedures provide *seamless mobility*
 - Enable real-time service continuity – PS Handover is a must-have for tight interworking of PS services
 - Three main scenarios were identified in 3GPP LTE-GSM workshop (January 2007):
 - Handover of voice services
 - CS voice
 - VoIP
 - Handover of packet data services
 - Handover of voice and packet data services

NOTE: GERAN/LTE PS Handover procedure will be specified as done for GERAN/UTRAN
NOTE: current loosest requirement (500ms) in 25.913 means PS Handover is required

Service Continuity – 3GPP Definitions

TS22.129

- **Service Continuity:** The means for maintaining active services during changes in the coverage areas or their characteristics **without**, as far as possible, **the user noticing**.
 - Note that Service Continuity can be achieved by handover, cell re-selection or other mechanisms.

TS22.258

- **Seamless:** **A user experience that is unaffected** by changes in the mechanisms used to provide services to a user
 - Note: The determination of whether something satisfies the requirement for being seamless or not is dependent on the user's (e.g., human end-user, protocol, application, etc.) perception of the service being received and not necessarily the technology used to provide the service.
- **Seamless Service:** Services provided across access systems and terminal capabilities. Provisioning of this service is continued between and within access systems and between terminals with minimal degradation and **no interruption in the service as seen by the user**, while adapting to the capabilities of each access system.

TS22.278:

- **Service Continuity:** The **uninterrupted user experience** of a service that is using an active communication (e.g. an ongoing voice call) when a UE undergoes a radio access technology change or a CS/PS domain change without, as far as possible, the user noticing the change.

Handover of Voice Services

Scope of the SR VCC in terms of mobility mechanism:

- Cell Reselections
- Handovers

Scope of the SR VCC in terms of service interruption requirement

- Requirements further confirmed in RAN2#57bis (EoMarch'07) by a number of operators (China Mobile, KPN, NTT DoCoMo, Orange, Sprint, T-Mobile, Telecom Italia, Vodafone : R2-071389 “LTE System Analysis of Control Plane and User Plane Latency and Handover Interruption Times”
 - *The interruption time during a handover of real-time services between E-UTRAN and GERAN is less than 300 msec*
 - *The interruption time during a handover of non real-time services between E-UTRAN and GERAN should be less than 500 msec*

What is understood with Voice Call Continuity?

- “**Maintain voice call quality**” which means that the quality would be maintained by allowing for a maximum service interruption of 300ms (current requirement/assumption) or less

Handover – Seamless Mobility

Solutions in SR VCC with several seconds of service interruptions are not allowing for voice service continuity following the definitions in 3GPP

Cell reselections – no need for SR VCC solutions – Mobility

Domain Transfer Requirement

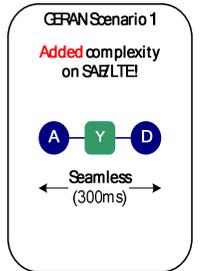
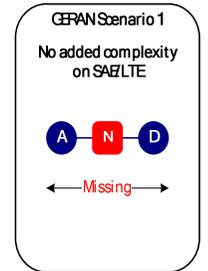
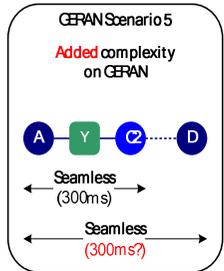
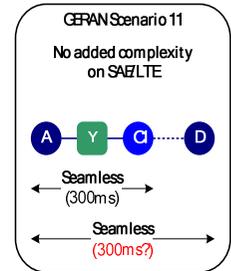
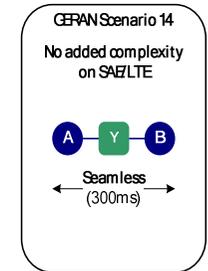
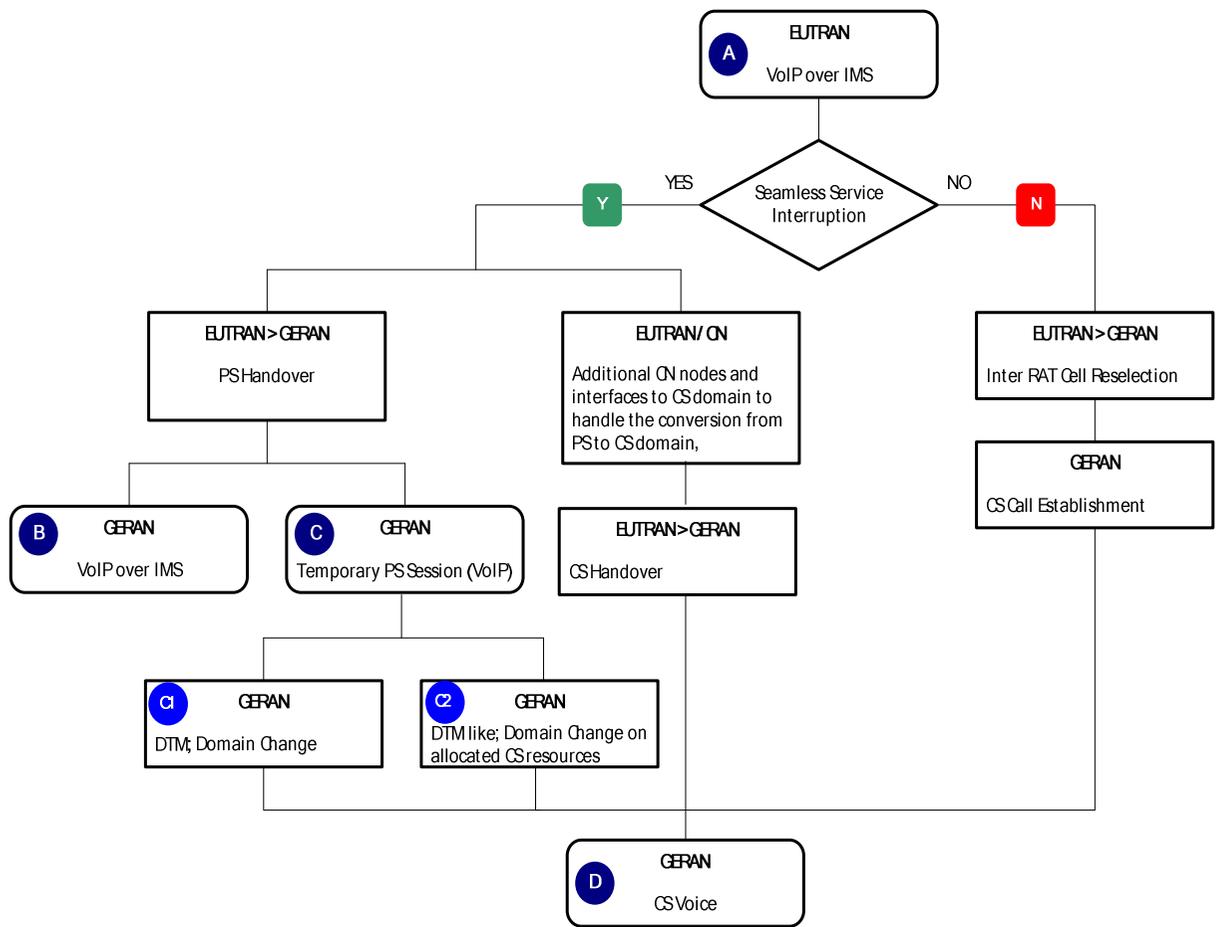
The requirements for domain change are missing

- In GERAN we have considered that the maximum service interruption is 300ms, however it needs to be determined whether the latency introduced by the domain change falls under inter-RAT handover requirements

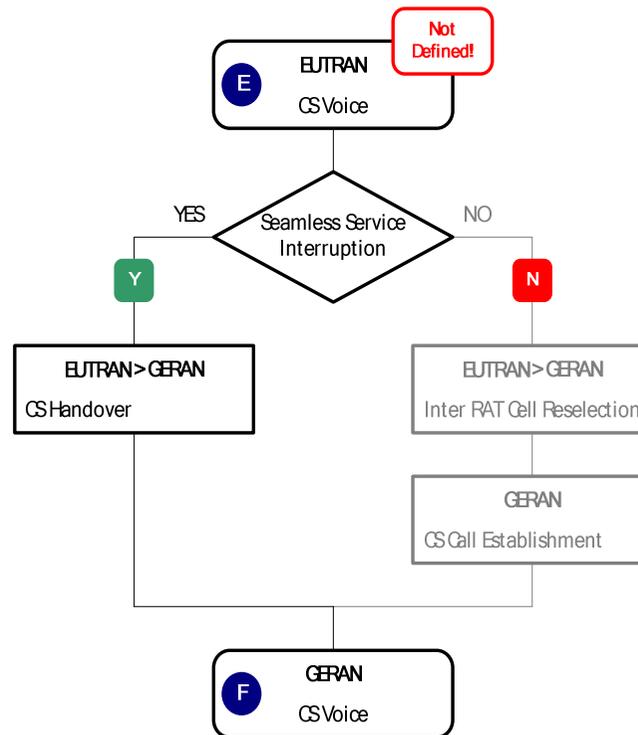
Copied from TS23.206:

- The following requirements are applicable to Domain Transfers in CS to IMS and IMS to CS directions
 - Initiation of the CS to IMS Domain Transfer procedures for an ongoing voice call may be based on radio condition.
 - Initiation of the IMS to CS Domain Transfer procedure for an ongoing voice call may be based on radio condition and IP connectivity to IMS
 - It shall be possible to support Domain Transfer for a roaming user
 - The **perceived service disruption should be minimized** from the end user's perspective.
 - **The Domain Transfer procedure latency should be minimized**
 - *In terms of service continuity is this latency part of the end-to-end delay requirement for real-time services or does it fall under inter-RAT handover requirement*
 - **Voice call quality should be maintained.** The number of transcoding stages introduced by the architecture should be minimized.
 - *This requirement should apply for inter-RAT change as well*
 - Consistent charging information across domains shall be provided when Domain Transfer is performed.

SR VCC – LTE/SAE PS domain only



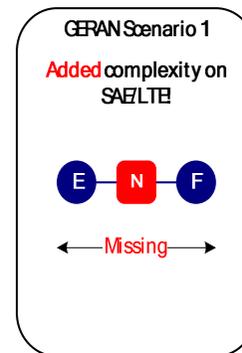
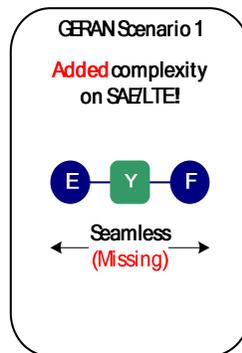
SR VCC - LTE/SAE CS domain



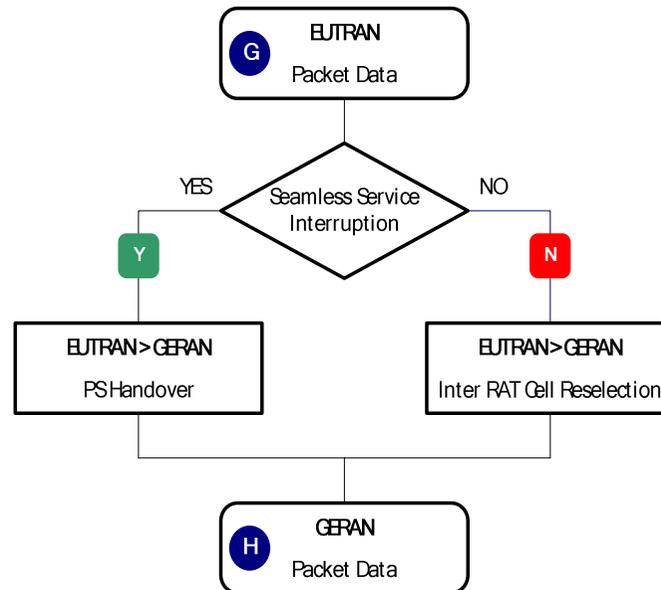
Grey: Unlikely scenario

Fundamental:
There is no CS domain in LTE/SAE!

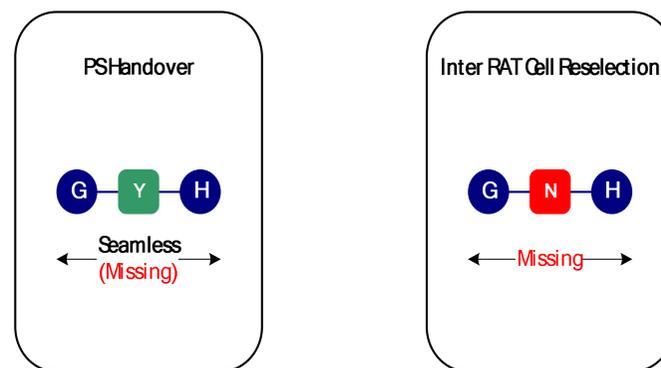
Proposal to reuse CS CC for voice in E-UTRAN -New SID in SA2



Other Packet Data



Single solution for voice and packet data!



Solutions on PS to CS approach targeting voice only require an additional PS to PS solution for packet data; at least two different solutions have to be supported!

Conclusions & Next Steps

SAE architecture support for ONE single solution for voice and packet data is being undermined as PS to CS path is the preferred approach in SA2

Note: Solutions based on PS to CS handover can only accommodate handover of voice services

Solution based on GERAN 11/14 scenario enables a single solution for all three handover scenarios:

- Voice (CS, VoIP)
 - Packet data
- Voice and packet data

Note: e.g. combinational VCC is such a solution.

Interworking preferred solutions for voice are:

- Redirection (see GP-07nnnn, S2-071931, next slide)
- Combinational VCC for seamless mobility (solution based on GERAN Scenario 11(14))

TSG GERAN should proceed with updating its specifications regardless of the work in SA2 for providing interworking with E-UTRAN (both directions)

- Cell re-selection mechanisms
- External NACC (studies are needed given SAE architecture)
- PS Handover

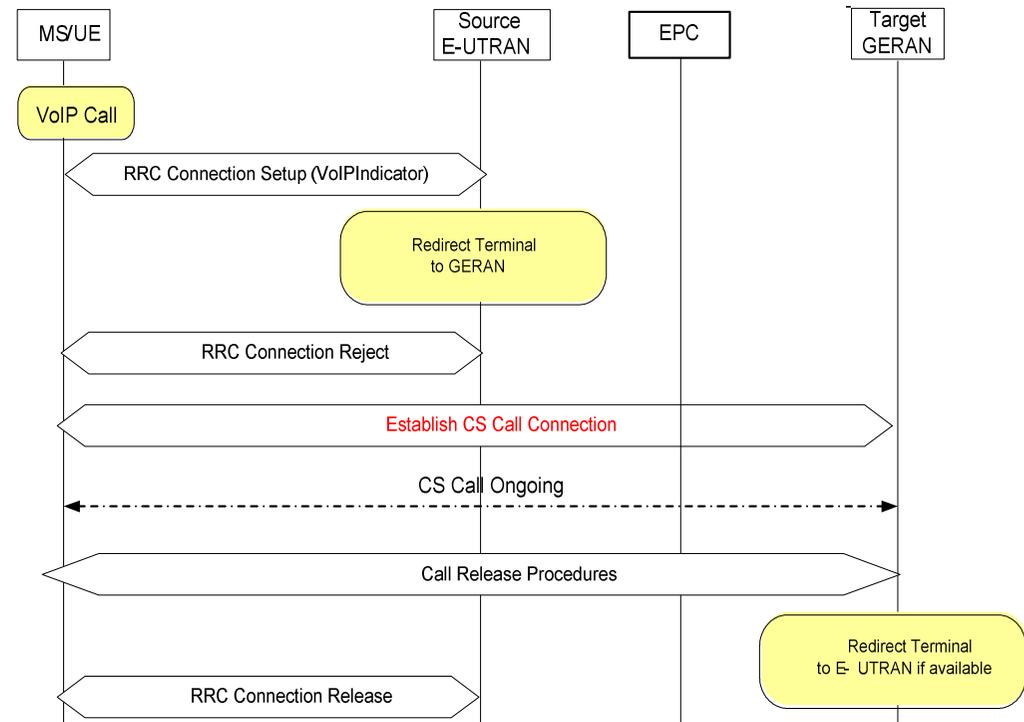
Redirection at call set-up

To avoid the problem of voice continuity during a voice call, the call could be redirected at call set-up similar to what already exists between e.g. GERAN and UTRAN (used in load sharing situation; video call redirection; ...)

Redirection needed from GERAN to LTE at call termination/release

Impact to be considered

- May have significant impact on service provisioning i.e. parallel packet data and voice is prevented (if no DTM supported) or potentially restricted (DTM)
- Redirection as a systematic mechanism for early LTE deployment might not be possible without investments in the target system: capacity, a bottleneck



Redirection as a possible first phase of LTE deployment (solution based on GERAN Scenario 1) – however impact must be well understood by operators and documented (see GP-070797)

Appendix – GERAN deployment scenarios

GERAN Features	CS HO	DTM (R99+)	EGPRS (EDGE) (R99+)	PS HO (Rel-6+)	DTM Enh. (Rel-6+)	VoIP (Rel-7+)	
Scenario 1				Must-have for tight interworking of packet services ($\leq 500\text{ms}$)			
Scenario 2							
Scenario 3							
Scenario 4							
Scenario 5							
Scenario 6							
Scenario 7							
Scenario 8							
Scenario 9							
Scenario 10							
Scenario 11							
Scenario 12	This scenario cannot exist – VoIP is built on top of EDGE						
Scenario 13							
Scenario 14							