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TECHNICAL SPECIFICATION

Satellite Earth Stations and Systems (SES); Family SL Satellite Radio Interface (Release 1); Part 4: Enhanced Services and Applications; Sub-part 1: Multiple Voice Services

Reference

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Contents

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Foreword

This Technical Specification (TS) has been produced by ETSI Technical Committee Satellite Earth Stations and Systems (SES).

The present document is part 4, sub-part 1 of a multi-part deliverable. Full details of the entire series can be found in ETSI TS 102 744-1-1 [4].

Modal verbs terminology

In the present document "**shall**", "**shall not**", "**should**", "**should not**", "**may**", "**need not**", "**will**", "**will not**", "**can**" and "**cannot**" are to be interpreted as described in clause 3.2 of the ETSI Drafting Rules (Verbal forms for the expression of provisions).

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Introduction

This multi-part deliverable (Release 1) defines a satellite radio interface that provides UMTS services to users of mobile terminals via geostationary (GEO) satellites in the frequency range 1 518,000 MHz to 1 559,000 MHz (downlink) and 1 626,500 MHz to 1 660,500 MHz and 1 668,000 MHz to 1 675,000 MHz (uplink).

1 Scope

The present document specifies the mandatory requirements for User Equipment (UE) implementing multiple voice communications via the Voice-over-IP (VoIP) service for the Family SL satellite radio interface.

2 References

2.1 Normative references

References are either specific (identified by date of publication and/or edition number or version number) or non-specific. For specific references, only the cited version applies. For non-specific references, the latest version of the reference document (including any amendments) applies.

Referenced documents which are not found to be publicly available in the expected location might be found at http://docbox.etsi.org/Reference.

NOTE: While any hyperlinks included in this clause were valid at the time of publication, ETSI cannot guarantee their long term validity.

The following referenced documents are necessary for the application of the present document.

[1] IETF RFC 3261 (2002): "SIP: Session Initiation Protocol", J. Rosenberg.

[2] IETF RFC 4733 (2006): "RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals", H. Schulzrinne.

[3] ETSI TS 102 744-1-4: "Satellite Earth Stations and Systems (SES); Family SL Satellite Radio Interface (Release 1); Part 1: General Specifications; Sub-part 4: Applicable External Specifications, Symbols and Abbreviations".

[4] ETSI TS 102 744-1-1: "Satellite Earth Stations and Systems (SES); Family SL Satellite Radio Interface (Release 1); Part 1: General Specifications; Sub-part 1: Services and Architectures".

[5] ETSI TS 124 008: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Mobile radio interface Layer 3 specification; Core network protocols; Stage 3 (3GPP TS 24.008)".

2.2 Informative references

References are either specific (identified by date of publication and/or edition number or version number) or non-specific. For specific references, only the cited version applies. For non-specific references, the latest version of the reference document (including any amendments) applies.

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The following referenced documents are not necessary for the application of the present document but they assist the user with regard to a particular subject area.

Not applicable.

3 Symbols and abbreviations

3.1 Symbols

For the purposes of the present document, the symbols given in ETSI TS 102 744-1-4 [3], clause 3 apply.

3.2 Abbreviations

For the purposes of the present document, the abbreviations given in ETSI TS 102 744-1-4 [3], clause 3 apply.

4 General

4.1 Overview

The Circuit Switched (CS) domain is designed to carry only one active voice call per subscriber identity at any time. For User Equipment (UE) that is shared among users, e.g. on ships or aircraft, it is desirable to overcome this limitation and to offer the ability to handle more than one concurrent voice call.

Additional voice calls can be provided by a Voice-over-IP (VoIP) service domain using the Packet Switched (PS) domain as its transport. The general principle of operation assumes that the first preference is to use the CS domain to set up a call. Only if the CS domain is unavailable (i.e. one voice call is already in progress between the user and another party), will any additional calls be set up using the VoIP domain.

The present document specifies the mandatory requirements for equipment implementing a User Agent in the VoIP service domain. This User Agent is essentially providing a Private Branch Exchange (PBX) in the mobile domain, as shown in Figure 4.1, and is therefore referred to as "Mobile PBX" throughout the present document.

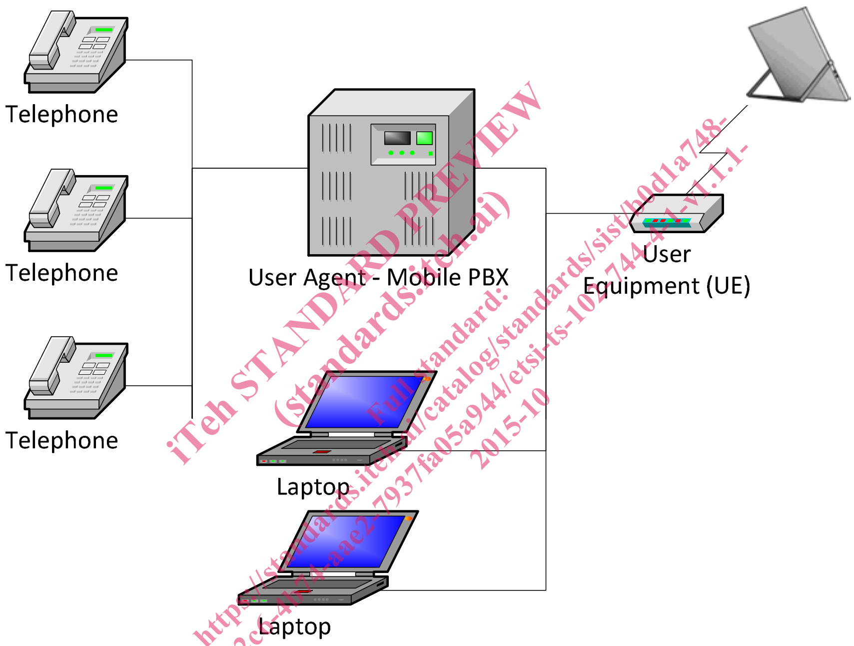

Figure 4.1: Equipment Configuration for support of Multiple Voice service

A Mobile PBX can be implemented either as an integral part of a UE or as external hardware that can be connected to an existing UE. In general, the requirements specified in the present document apply to both implementations unless indicated otherwise.

4.2 System architecture

The solution described in the present document is based upon the integration of a VoIP domain containing infrastructure operating in tandem with the Circuit Switched domain and operating via the Packet Switched Domain. A simplified representation of the overall system architecture is represented in Figure 4.2, where the VoIP domain is functionally provided by a VoIP server.

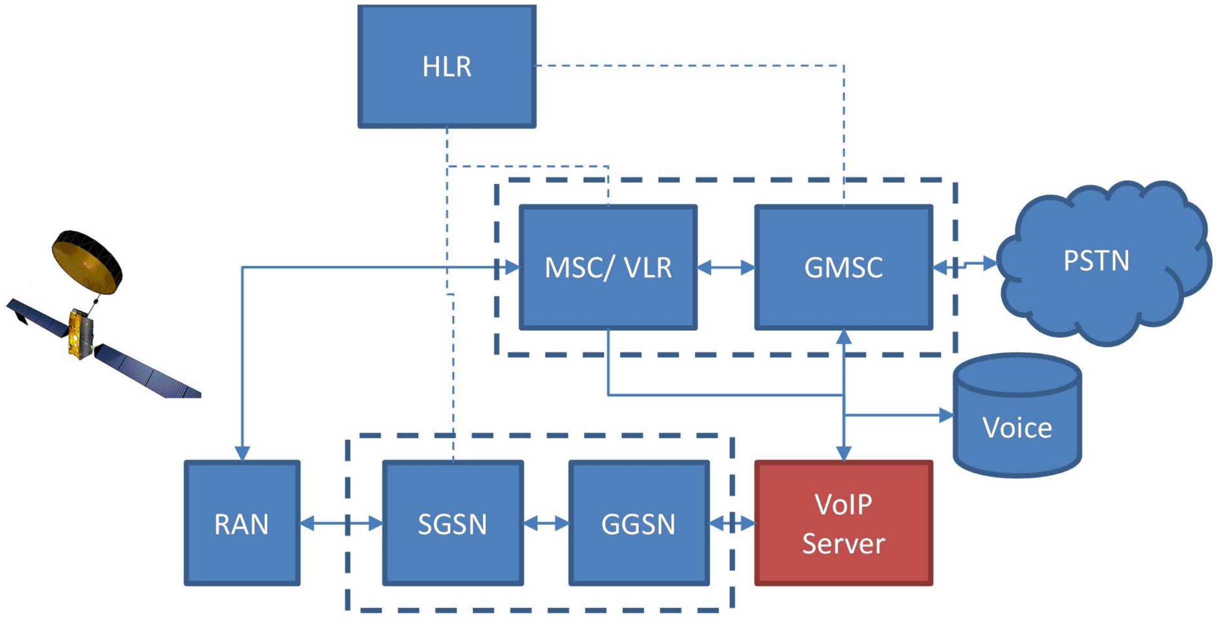

Figure 4.2: Overall System Architecture for support of Multiple Voice service

A new VoIP Domain, provided via a VoIP server, is introduced into the Family SL ground segment, and provides an interface between the existing Circuit Switched domain and the Packet Switched domain for transport of multiple voice calls when the single CS voice circuit towards a mobile terminal is busy.

The interface with the PSTN is handled by the existing Circuit Switched domain for both mobile terminated and mobile originated calls. The existing voice mail system will be used to support per-MSISDN voice mail.

Multiple MSISDN numbers are allocated to the subscriber. These numbers may be allocated to individual handsets by configuration of the mobile terminal or the external VoIP PBX. Alternatively functionality may be provided in the mobile terminal or VoIP PBX to determine which handset to alert upon Mobile Terminated call initiation utilizing the primary MSISDN.

4.3 Access to the VoIP domain

The Mobile Terminal registers with the Radio Access Network and then initiates an Attach procedure towards both the CS and PS domains. During these Attach procedures the MT is authenticated, and Security Mode procedures are initiated to secure the signalling and user-plane between the Radio Access Network and the Mobile Terminal.

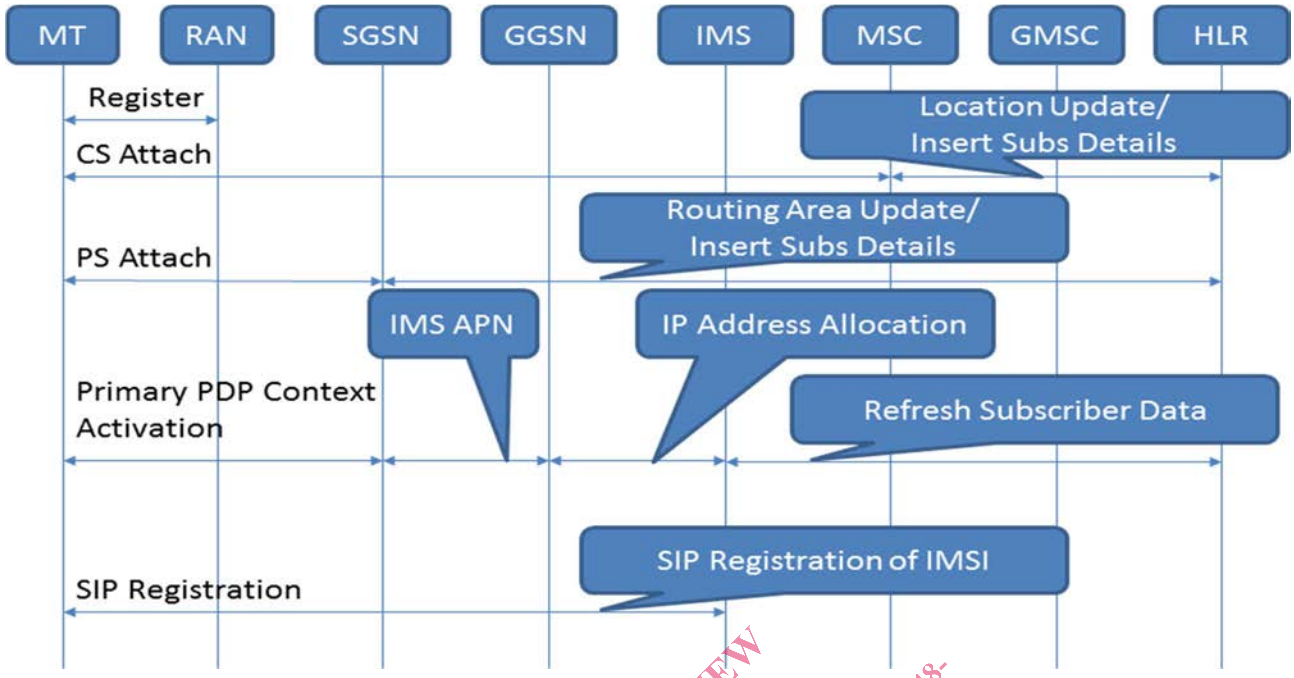

Figure 4.3: Simplified Presentation of Initial Access Procedure

Conditional on successful CS and PS attach procedures , the mobile terminal initiates a background Primary PDP context towards the Inmarsat VoIP APN, which is authorized by the HLR if the subscriber is granted access to Multiple Voice services. Conditional on the Mobile Terminal subscriber being authorized to access VoIP domain infrastructure, an IP address is allocated to the mobile terminal and the PDP context activated.

The Mobile Terminal then performs a DNS lookup to obtain the IP address of the SIP server within the VoIP domain (the fully qualified domain name of the SIP server will be configured in the Mobile Terminal). The Mobile Terminal then initiates a SIP registration procedure towards the SIP server.

5 Mobile PBX Requirements

5.1 General Requirements

5.1.1 Number of Concurrent Calls

The Mobile PBX shall support a minimum of four concurrent voice calls (one call placed via the CS domain and an additional three placed through the PS domain). It should support up to nine concurrent voice calls.

NOTE: The maximum number of concurrent voice calls supported is determined by the subscribers' service package and is enforced by the network.

5.1.2 Support for Direct Inward Dialling

Additional MSISDNs (AMISDNs) may be allocated to a UE network subscription to provide support for Direct Inward Dialling (DID) from another network to individual telephone handsets connected to the Mobile PBX. These DID numbers shall be provided to the subscriber when they request a multiple voice service package from the satellite network service provider.

The Mobile PBX may optionally support telephone handsets that are not associated with a DID number. Such handsets would typically be used for outbound calls only.

5.1.3 Support for Short Code Dialling

The Mobile PBX shall support Short Code Dialling in both service domains.