CALLUP Call back Solution
Product Description

CALLUP Callback Platform

Technical Product Description

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1 Overview

This document provides a detailed technical description of CALLUP's callback platform, including its features, capabilities, architecture etc.

CALLUP's callback platform provides mobile operator to keep the bulk of income from roaming calls in house. When a subscriber is roaming abroad and makes direct call from his mobile, the operator abroad sees most of the income from this call. With callback, the subscriber is (transparently) getting an incoming call, and can be billed less than what he would pay if he called directly.

CALLUP's callback services contain the following:

- **Hardware:**
    - **State of the art** HP industrial servers featuring Intel processors and Intel/Dialogic E1 and SS7 hardware.
    - **Virtual Machine Ready**: An all IP system can be run on virtual machines, save precious HW costs, power consumption and floor space!
- **2G/3G STK Client Application**: Catch the outgoing call with the highest success rates, OTA configurable, runs on any card type and supports both SMS and USSD interconnection with the server.
- **Compatibility with all major Switches and HLRs:** The platform is compatible with all major Switches and facilitates seamless migration from one Switch vendor to another.
- **Unlimited scalability**. A system with 1000 BHCA can be upgraded to a system with 3,000,000 BHCA in one rack, either TDM or SIP.
- **Seamless integration** to other billing/IT systems
- **Minimal training**: Since the system utilizes standard components, a few hours of training is normally sufficient
- **Unique remote support** hardware enabling complete control and monitoring of the system from anywhere on the network
- **Optional Service Creation Environment**: Easily create drag and drop services using the web based GUI for flow control.

2   Main Features

- STK application for automatic diversion of outgoing calls to the callback system
- OTA configurable applet (for example, enable/disable per country/network)
- Both USSD and SMS support (client and server side)
- Internal USSD gateway
- Allow service only from specific countries/network (enforce with SRI)
- Allow only prepaid or only post paid subscribers
- Smart configuration mechanism, for overcoming device comparability issues and USSD restriction issues
- Per-TAC, MCC and MNC policies
- User feedback and status, over USSD or SMS
- Configurable dialed number validation program – allow only valid numbers through the system
- Web based GUI for the management of the platform
- TDM (MTP, ISUP) or IP (SIGTRAN, SIP) for signaling and voice

3   User Experience

The user experience of the USSD Callback service is described in the scenario below.

1. The A-party (which is roaming abroad) dials a number normally (as if the number was dialed in the home country).

2. The call is disconnected, and a proper text message is displayed to the user telling to wait for an incoming call.
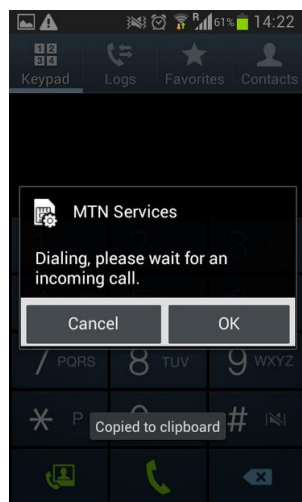
3. After a few seconds, the A-party receives an incoming call with a special CLI. This is the Callback service CLI which is predefined by the HPMN.

4. The A-party answers the call and hears the ring back tone of the call to the B-party.

5. Meanwhile, the B-party's phone is ringing and displaying the CLI of the A-party.

6. If the B-party answers the call, the 2 parties can start talking.

7. If the call is not answered for some reason (busy, no answer, unavailable), then the A-party hears the response on the other side and hangs up.

4   Architecture

4.1   Diagram
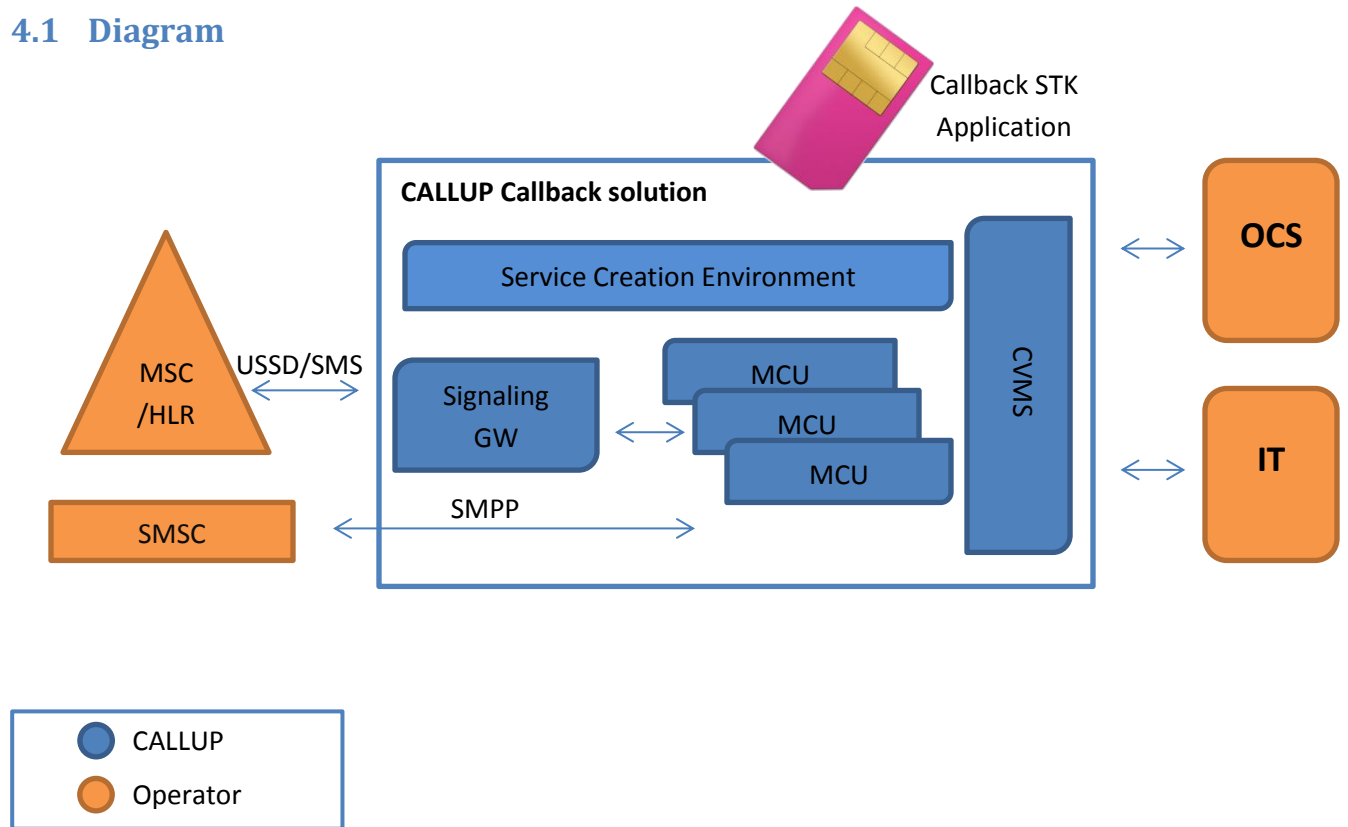
5   Modules

CALLUP's callback solution is comprised from the following modules:

5.1   Network and media module

This module is the gateway to the SS7 network, and provides the required GSM MAP interface towards the network (for accepting and sending USSD messages), and the SMPP interface (for SMS interconnection).

For SS7 interconnection, both TDM and IP are supported:
- Over TDM using E1 or T1 interface
- Over SIGTRAN (SCTP/M3UA)

The module is relying on the Dialogic set of network and media products, for providing the most robust and network proven signaling and media processing capabilities. These capabilities are achieved in various ways, depending on the scale and the technology used:

- Dedicated redundant SIU (Signaling gateway) servers, for high load configurations
- Signaling E1/T1 cards on the MCU machine
- No special hardware required for SIGTRAN

All of the above complies with the latest ITU-T recommendations and ETSI standards.

5.1.1   SS7

The SS7 integration is done using either SIUs for large scale or SS7 boards in smaller scale solutions. The 2 Intel SIU signaling servers are installed with ISUP protocol stack. These servers are configured in a redundant fault tolerant configuration that requires 2 SS7 links but only 1 point code. The SIU units communicate with the MCU units via TCP/IP. The SIU complies with ITU-T recommendations and ETSI standards.

Alternatively, a single Dialogic DSI Quad E1 cards may be installed in each of the MCU units. The SS7 signaling and E1 voice may be combined on a single E1 cable.

5.2   Voice (IVR) module

This module is the machine that actually creates calls to the original subscribers. Dedicated lines (the number of which is determined in the dimensioning phase of the project) are used to make calls to the roaming subscribers. Detailed, customizable CDRs are generated per each call, for charging purposes. The IVR voice trunks can be either TDM based, with voice over E1s and ISUP for call control, or IP based, with RTP for voice and SIP for call control.

5.3   Application module

This module executes the various call flows, which is the logic of the application. The application handles requests from various sources and of various types, and handles them according to the pre-defined logic.

5.4   CVMS

CALLUP's CVMS (CALLUP Versatile Management System) is the web based GUI service management console. It enables stop/start of services, monitoring the health and performance of services, doing bulk operations on services and so on. All of CALLUP's software is managed from this system.

This means you can constantly monitor each transaction, restart services, monitor SS7 status, view logs and CDRs etc.

The CVMS is also the module that enables SNMP traps towards the operator's central NOC. When integrating the IVR to the operator's network, CALLUP's MIB file is shared, together with a description of all the alarms, their description and severity, so that the external NOC can monitor the system.

5.5   STK Application

The applet captures a roaming call attempt, converts it to USSD or SMS request to the server, which in turn calls back the caller and the original recipient. The Applet can be fine-tuned OTA to different operating modes.

5.5.1   Overview of the STK application's features:

- Automatic capture of an outgoing call (seamless), and conversion to a USSD or SMS string
- Automatic operation when roaming
  - Determination of "roaming" state configurable, can match any business requirements
- Rich configuration file which provides flexibility for the operator
- Support both USSD or SMS mode operation
- Smart selection of USSD/SMS based on MCC/MNC or device type - "device sensitivity" premium feature
- Configurable strings in any language
- OTA configurable (requires an OTA system)

5.6   The applet registers to the following events:

- 'PROFILE DOWNLOAD'
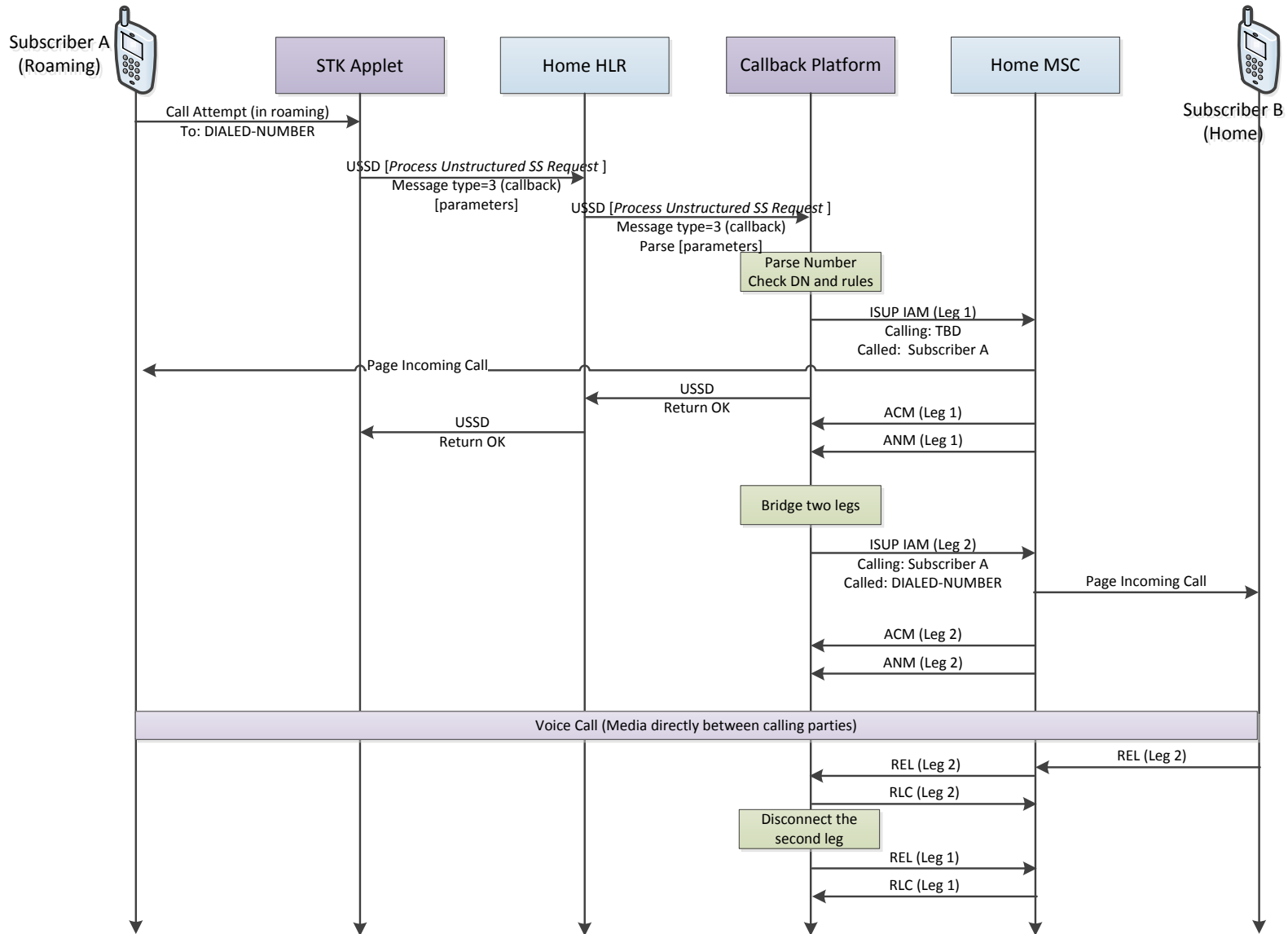- 'STATUS'
- 'CALL CONTROL'

5.7   The applet will use the following proactive commands:

- 'PROVIDE LOCAL INFORMATION'
- 'SEND USSD'
- 'SEND SMS'
- 'SELECT ITEM'
- 'DISPLAY TEXT'
- 'SET POLL INTERVAL'
- 'SET UP MENU'

6   Call Flows

6.1   Making a call to a home subscriber while roaming (ISUP)

Successful USSD Callback Call - ISUP

Subscriber A
(Roaming)

STK Applet

Home HLR

Callback Platform

Home MSC

Subscriber B
(Home)

Call Attempt (in roaming)
To: DIALED-NUMBER

USSD [*Process Unstructured SS Request*]
Message type=3 (callback)
[parameters]

USSD [*Process Unstructured SS Request*]
Message type=3 (callback)
Parse [parameters]

Parse Number
Check DN and rules

ISUP IAM (Leg 1)
Calling: TBD
Called: Subscriber A

Page Incoming Call

USSD
Return OK

ACM (Leg 1)

USSD
Return OK

ANM (Leg 1)

Bridge two legs

ISUP IAM (Leg 2)
Calling: Subscriber A
Called: DIALED-NUMBER

Page Incoming Call

ACM (Leg 2)

ANM (Leg 2)

Voice Call (Media directly between calling parties)

REL (Leg 2)

REL (Leg 2)

RLC (Leg 2)

Disconnect the
second leg

REL (Leg 1)

RLC (Leg 1)

6.2 Making a call to a home subscriber while roaming (ISUP) - Call B first (with media)
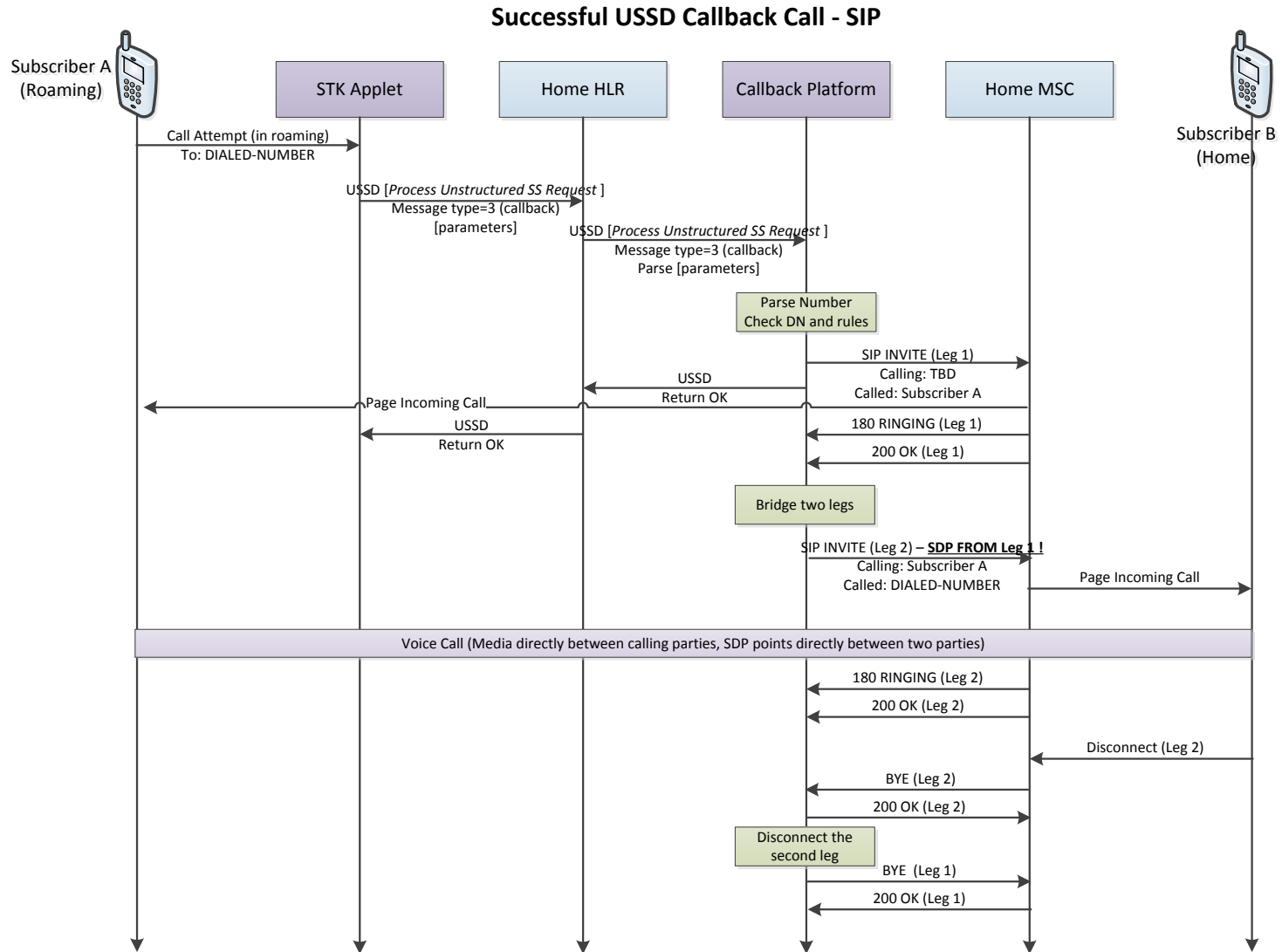
Successful USSD Callback Call – ISUP, No STK Applet

6.3   Making a call to a home subscriber while roaming (SIP)
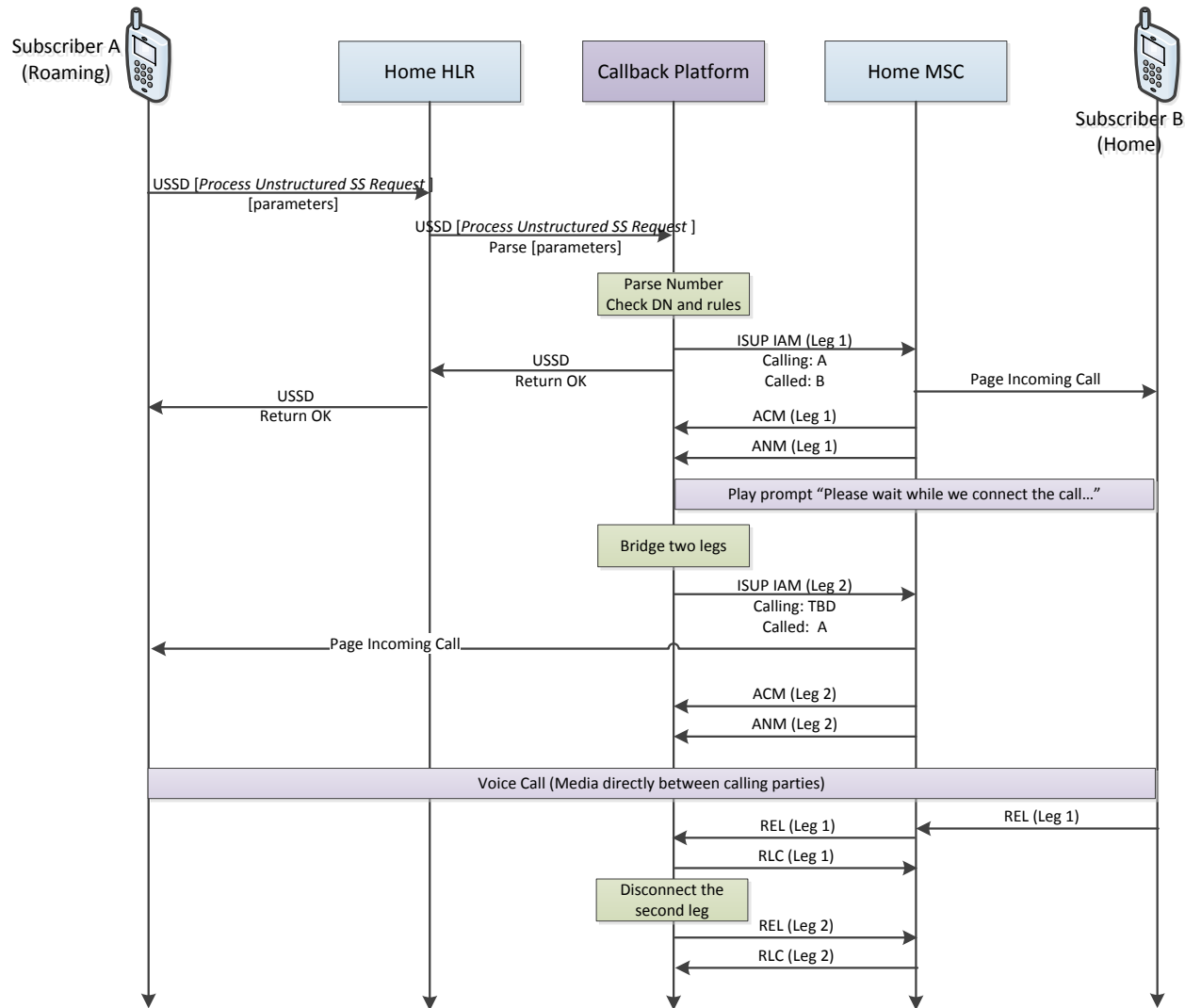
Successful USSD Callback Call - SIP

6.4   STK application power up/on/off flow

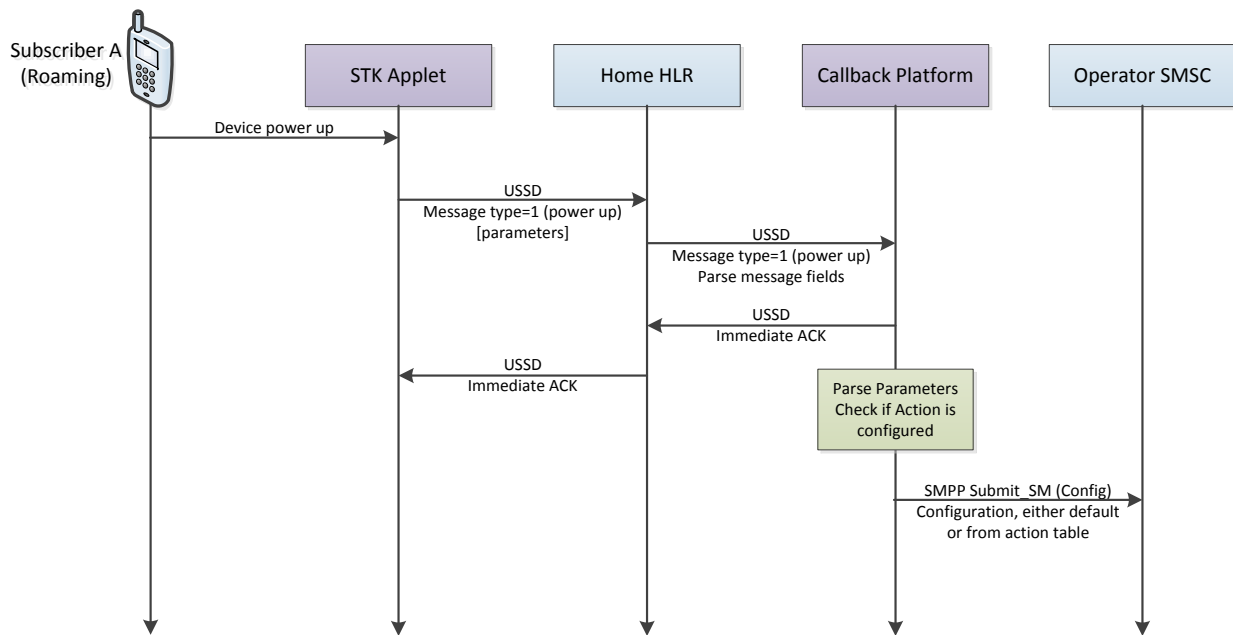**STK Power Up/On/Off Event Flow**

Figure 1: Per-TAC/MCC/MNC Configuration of applet

7   Billing Support

7.1 Postpaid Support

For support of postpaid billing, the USSD Callback solution provides CDR files, which contain information on all callback transactions. CDR files can be transferred to the operator every 15 minutes.

7.2 Prepaid Support

For support of prepaid subscribers, the USSD Callback solution relies on the HPMN MSC capabilities for charging the A-party for the first leg and the second leg.

Charging for Leg-1 (from Callback platform to the A-party): Since the A-party is roaming abroad, the HPMN MSC must charge every MT call that is sent to him. In this case, the HPMN MSC will send an MT trigger to the Prepaid SCP to check for credit and allow charging.

Charging for Leg-2 (from Callback platform to the B-party): Assuming that the B-party is currently in the Home country, there will not be an MT trigger to the SCP. In this case, the Callback platform will add a predefined prefix to the Called Party Number. This prefix will allow the HPMN MSC to send a network trigger to the Prepaid SCP in order to charge the A-party for this call leg.

7.3   Barring Support

For support of call barring for postpaid subscribers, the USSD Callback solution relies on the HPMN MSC capabilities. Since the A-party is roaming abroad, the HPMN MSC must check barring status for every MT call that is sent to him. If the A-party is barred, then the MSC will not allow Leg-1 to be formed and the USSD Callback solution will abort the session.

8   Redundancy and Fault Recovery

8.1   Signaling and media gateways

For high scale solutions, two signaling servers (Dialogic SIU Gx1) are provided which service all MCU units. Each MCU connects to the signaling servers (load balancing) and one server will automatically take over in the event that the 2nd server fails with no loss of service.

8.2   Message Control Units (MCU)

CALLUP callback solution features N+1 MCU redundancy. All MCUs operate simultaneously (Load Balancing). The system will continue functioning in the event that one MCU unit fails with no loss of capacity.

8.3   Ethernet Switches

Two 1Gb/10Gb Ethernet switches are provided where all servers are connected to both Switches. The Ethernet switches are both redundant (one switch can fail) and load balancing (each server can transmit on 2 Ethernet interfaces). Ethernet teaming is defined on 2 NICS on each server with transmit load balancing such that the server transmits on both NICs (load balancing) and does not lose connection with the switch when 1 link fails (redundancy)

8.4   Server Hardware Redundancy

All servers have redundant power supplies and fans. RAID 1 disks with automatic hot spare ensure that there will be no loss of service in the event that a system disk fails. On-line spare disks automatically replace a failed system disk until it is replaced. The SDU servers have redundant Hot Swap fiber controllers. Hot swap spare memory ensures that the server will not fail even if a memory bank fails.

9   Scalability

Additional E1 lines may be provided simply by adding E1 cards in the 2 MCU units or adding additional MCU units (up to 10 MCUs).

All of the application and signaling servers, as well as the Ethernet switches, are available in both AC and DC (-48V) power.

Various backup solutions, including tape solutions (such as SDLT and LTO auto changers) or cloud based backup solution, can also be provided by CALLUP.