GSM Voice Mail Service TDM Call Control

Ebby Wiselyn Jeyapaul
Abstract

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*Ebby Wiselyn Jeyapaul*

The Voice Mail Service (VMS) enables forwarding of calls to a dedicated Voice Mail Server (VMS) on behalf of the call receiving subscriber during certain conditions such as 'busy subscriber', 'no answer', 'always', etc.

The standardization forum 3GPP has specified the Global System for Mobile communication (GSM) while the standardization forum ITU-T has specified the Integrated Services Digital Networks (ISDN) User Part (ISUP) call control protocol. Both of these standards rely on the use of Time Division Multiplex (TDM) as a media bearer and SS7 as signalling bearer, where both bearers require use of very expensive telecom-specific hardware.

The thesis proposes the solution to use RTP as media bearer and IP as signalling bearer towards the handset in GSM and only use TDM as media bearer and SS7 as signalling bearer towards the VMS.

The thesis demonstrates the feasibility and the advantages provided, by creating an implementation in Erlang/OTP and testing it to check if it confirms to the specification.
Acknowledgements

This master thesis would not have been possible without the guidance and help of my supervisor Lars Kari, and my reviewer Ping Wu. I would like to thank all my friends and family who supported me during my stay and study in Uppsala University.
**Glossary**

<table>
<thead>
<tr>
<th><strong>Abbreviation</strong></th>
<th><strong>Description</strong></th>
</tr>
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<tbody>
<tr>
<td>3GPP</td>
<td>The 3rd Generation Partnership Project</td>
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<tr>
<td>Abis</td>
<td>Abis interface, Radio Signalling Link</td>
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<td>A-law</td>
<td>A-law algorithm</td>
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<td>AN</td>
<td>Access Network</td>
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<td>BSC</td>
<td>Base Station Controller</td>
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<td>BSS</td>
<td>Base Station Subsystem</td>
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<td>BSSMAP</td>
<td>Base Station System Management Application Sub-part</td>
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<td>BTS</td>
<td>Base Transceiver Station</td>
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<td>CAMEL</td>
<td>Customized Applications for Mobile network Enhanced Logic</td>
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<td>CEPT</td>
<td>European Conference of Postal and Telecommunications Administrations</td>
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<tr>
<td>CF</td>
<td>Call Forwarding</td>
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<td>CFB</td>
<td>Call Forwarding Busy</td>
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<td>CFU</td>
<td>Call Forwarding Unconditional</td>
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<td>CFNRc</td>
<td>Call Forwarding Not Reachable</td>
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<td>CFNRy</td>
<td>Call Forwarding No Reply</td>
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<td>CIC</td>
<td>Circuit Identification Code</td>
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<td>CN</td>
<td>Core Network</td>
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<td>CS</td>
<td>Connection Service</td>
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<td>DPC</td>
<td>Destination Point Code</td>
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<td>DTAP</td>
<td>Direct Transfer Application Part</td>
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<td>DTMF</td>
<td>Dual Tone Multi-frequency</td>
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<td>E-Mail</td>
<td>Electronic Mail</td>
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<td>E1</td>
<td>European-carrier 1</td>
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<td>EFR</td>
<td>Enhanced Full Rate</td>
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<td>ETSI</td>
<td>European Telecommunications Standards Institute</td>
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<td>G-MSC</td>
<td>Gateway Mobile-service Switching Centre</td>
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<td>GSM</td>
<td>Global System for Mobile Communication</td>
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<td>GTH</td>
<td>Generic Telecom Hardware</td>
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<td>HLR</td>
<td>Home Location Register</td>
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<td>IAM</td>
<td>Initial Address Message</td>
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<td>Acronym</td>
<td>Description</td>
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<tr>
<td>IMSI</td>
<td>International Mobile Subscriber Identity</td>
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<td>INAP</td>
<td>Intelligent Network Application Part</td>
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<td>IP</td>
<td>Internet Protocol</td>
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<td>ISDN</td>
<td>Integrated Services Digital Network</td>
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<td>ISUP</td>
<td>ISDN User Part</td>
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<td>ITU-T</td>
<td>The ITU Telecommunication Standardization Sector</td>
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<td>M3UA</td>
<td>MTP Level3 User Adaptation Layer</td>
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<td>MA</td>
<td>Mobile Arts</td>
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<td>MAP</td>
<td>Mobile Application Part</td>
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<td>MAF</td>
<td>Mobile Additional Function</td>
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<td>MCC</td>
<td>Mobile Country Code</td>
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<td>MNC</td>
<td>Mobile Network Code</td>
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<td>ME</td>
<td>Mobile Equipment</td>
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<td>MG</td>
<td>Media Gateway</td>
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<td>MGC</td>
<td>Media Gateway Controller</td>
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<td>MMS</td>
<td>Multimedia Messaging Service</td>
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<td>MS</td>
<td>Mobile Station</td>
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<td>MSC</td>
<td>Mobile-service Switching Centre</td>
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<tr>
<td>MSISDN</td>
<td>Mobile Subscriber Integrated Services Digital Network Number</td>
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<td>MSU</td>
<td>Message Signalling Unit</td>
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<td>MTP</td>
<td>Message Transfer Part</td>
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<td>MUS</td>
<td>Mobile User Service</td>
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<tr>
<td>NSS</td>
<td>Network Switching Subsystem</td>
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<td>OPC</td>
<td>Originating Point Code</td>
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<td>PLMN</td>
<td>Public Land Mobile Network</td>
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<td>POTS</td>
<td>Plain Old Telephony Systems</td>
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<td>PSTN</td>
<td>Public Switch Telephone Network</td>
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<td>RSL</td>
<td>Radio Signalling Link</td>
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<td>RTP</td>
<td>Real-time Transport Protocol</td>
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<td>SCP</td>
<td>Service Control Point</td>
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<td>SCCP</td>
<td>Signalling Connection Control Part</td>
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<td>SCTP</td>
<td>Stream Control Transmission Protocol</td>
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<td>SIGTRAN</td>
<td>Signalling Transport</td>
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<td>SIM</td>
<td>Subscriber Identity Module</td>
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<tr>
<td>SMS</td>
<td>Short Message Service</td>
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<td>SS</td>
<td>Supplementary Service</td>
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<td>SS7</td>
<td>Signalling System 7</td>
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<td>TCAP</td>
<td>Transmission Capability Application Part</td>
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<td>TDM</td>
<td>Time Division Multiplexing</td>
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<td>T-MSC</td>
<td>Terminating Mobile-service Switching Centre</td>
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<tr>
<td>Abbreviation</td>
<td>Description</td>
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<tr>
<td>TM</td>
<td>Transit Module</td>
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<tr>
<td>TMSI</td>
<td>Temporary Mobile Subscriber Identity</td>
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<tr>
<td>NDUB</td>
<td>Network Determined User Busy</td>
</tr>
<tr>
<td>UDUB</td>
<td>User Determined User Busy</td>
</tr>
<tr>
<td>USSD</td>
<td>Unstructured Supplementary Service Data</td>
</tr>
<tr>
<td>V-MSC</td>
<td>Visiting Mobile-service Switching Centre</td>
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<td>VLR</td>
<td>Visitor Location Register</td>
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<td>VM</td>
<td>Voice Mail</td>
</tr>
<tr>
<td>VMS</td>
<td>Voice Mail Service</td>
</tr>
</tbody>
</table>
# Contents

1 Introduction ........................................... 1
   1.1 Background ......................................... 2
   1.2 Motivation .......................................... 5
   1.3 Purposes And Goals .................................. 6
   1.4 Thesis Structure ..................................... 7

2 GSM Overview and SS7 Protocols ......................... 8
   2.1 GSM Overview ....................................... 9
      2.1.1 Access Network ................................ 10
      2.1.2 Core Network .................................. 11
      2.1.3 Location Procedures ........................... 13
      2.1.4 Interfaces ..................................... 13
   2.2 Signalling System 7 Protocols Overview ............ 15
      2.2.1 M3UA ............................................ 17
      2.2.2 Integrated Services Digital Network (ISDN) User Part ... 17
      2.2.3 Mobile Application Part (MAP) .................. 19
   2.3 GSM Call Service .................................... 21

3 GSM Voice Mail Service and its Call Control .......... 22
   3.1 Use Cases .......................................... 23
   3.2 System Capabilities .................................. 26
   3.3 Media Control Integration ............................ 28

4 Design .................................................. 29
   4.1 Architecture ........................................ 31
   4.2 Components ......................................... 34
      4.2.1 Mobile User Service ............................ 35
      4.2.2 VLR .............................................. 37
      4.2.3 MAP Handlers .................................... 38
      4.2.4 Transit Module .................................. 39
      4.2.5 Connection Service .............................. 39
4.3 Call Control Interfaces ........................................... 40
  4.3.1 Map Handler Signals ........................................ 40
  4.3.2 VLR Interface ............................................... 41
  4.3.3 HLR Interface ............................................... 41
  4.3.4 MUS Interface ............................................... 41
  4.3.5 Connection Server Interface ............................... 41
  4.3.6 ISUP Access Handler ↔ Tieto ISUP stack .................... 41
  4.3.7 Transit Module ↔ ISUP access handler ....................... 42
4.4 Handling Bearer Differences .................................... 43
  4.4.1 ISUP Mapping ............................................... 43

5 Implementation And Test ........................................... 45
  5.1 Methodology .................................................... 46
  5.2 Environment Setup ............................................ 46
  5.3 Implementation Groups ...................................... 47
  5.4 Design Phases ................................................ 48
  5.5 Testing ......................................................... 49

6 Discussions & Conclusions ......................................... 51
  6.1 Features ....................................................... 52
  6.2 Performance .................................................. 52
  6.3 Security ....................................................... 52
  6.4 Cost .......................................................... 52
  6.5 Conclusions ................................................... 53

7 Future Work ......................................................... 54
Chapter 1

Introduction
1.1 Background

The first generation cellular systems had a diverse growth in the early 1980s, each development was incompatible with other existing ones at that time. Global System for Mobile Communication (GSM) is a standard set developed by European Telecommunications Standards Institute (ETSI) to describe protocols for second generation (2G) cellular networks (that are digital). It was formed by the European Conference of Postal and Telecommunications Administrations (CEPT) to study and develop a pan-European public land mobile system.

Phase 1 of the GSM was published in 1990, and resumed with the service the following year. GSM evolved from a European standard to being operational in more than 100 countries around the world. After a span of over four years, there were over one million subscribers worldwide, having had a gigantic expansion of over 50 million subscribers in the next 3 years.

GSM provides compatibility with ISDN services, and bearer signalling, teleservice and supplementary services, with telephony being the heart of all the services, using digital encoding and transmission of speech.

The telephony services use the capabilities of a bearer service to transport data, which defines the resource capabilities required for the call. Voice calls are the basic services which are supported with Full-rate speech with 13kbps along with emergency calls.

Short messaging services allows subscribers to send and receive text messages on the GSM handset. Provided above the bearer and teleservices are the supplementary services, offering number identification, call forwarding services, call barring, call hold, call waiting and conference services.

External systems in the GSM network like the Voice Mail Service (VMS) provide facility for subscribers to leave their message in the recipient’s mailbox. This gives the subscriber more control to manage their inflow of calls, and user absence. Voice mail systems are usually made compatible with GSM, UMTS, and CDMA networks.

Signalling System 7 (SS7) is a set of telephony signalling protocols(messages) which are used to setup most of the world’s public switched telephone network telephone calls. SS7 is mainly used to set up, supervise and tear down telephone calls. It is also employed for Short Message Service (SMS), number translation services, local number portability, and other market services. SS7 is the core signalling backbone.
of telephony networks around the world, used by a variety of networks including GSM, ISDN, Public Switch Telephone Network (PSTN), Intelligent Networking, etc.

The following figure (Fig 1.1) presents an overview of a GSM Network and its connection to ISDN and Plain Old Telephony Systems (POTS) through SS7 signalling bearer. The details on GSM and SS7 are given in chapter 2.

![GSM and SS7 Network](image)

**Figure 1.1: SS7 and GSM Network. [1]**

The standardization forum The 3rd Generation Partnership Project (3GPP) has specified the GSM while the standardization forum The ITU Telecommunication Standardization Sector (ITU-T) has specified the ISDN User Part (ISUP) call control protocol. Both of these standards rely on the use of Time Division Multiplexing (TDM) as a media bearer and SS7 as signalling bearer, where both bearer require use of very expensive telecom-specific hardware. A cheaper option would be to always use Real-time Transport Protocol (RTP) as media bearer and
Internet Protocol (IP) as signalling bearer towards the handset in GSM and only use TDM as media bearer and SS7 as signalling bearer toward the VMS.

Thus, it would be interesting to investigate the possibility to implement the call control part of a Voice Mail (VM) call forwarding service in an existing GSM call service environment using RTP as media bearer and IP as signalling bearer toward the GSM handset and using TDM as media bearer and ISUP/SS7 as signalling bearer towards the VMS application server.

To run SS7 protocols over IP, one can use MTP Level3 User Adaptation Layer (M3UA) adaptation [14], which was studied during the Project CS 2011 course in Uppsala University, and used to to develop the GSM Call Service. *GSM Call Service* provides a vendor independent, low cost, IP based solution, for calls, subscriber management and additional GSM services.

The existing *GSM Call Service* environment is provided by Uppsala University’s Computer Science project course 2011 including an existing GSM Home Location Register (HLR) provided by Mobile Arts where SS7 shall be used as signalling bearer between the Mobile-service Switching Centre (MSC)/Visitor Location Register (VLR) and the Home Location Register (HLR). The VMS application server is also provided by Mobile Arts.
1.2 Motivation

Since both GSM and ISUP specify signalling and media bearers that rely on expensive telecom hardware, there is an increasing need for cheaper alternatives that can be used as bearers for both signalling and media. This has a cost advantage if traditional SS7 links are replaced by IP connections.

The motivation for the thesis is to find ways to develop frameworks that can enable GSM to be used under academic environments, and for creating test frameworks to test existing GSM systems using ISUP/IP as signalling bearer, and RTP as media bearer instead of expensive telecom-specific hardware. The thesis also attempts to add the body of knowledge to the ongoing research on M3UA systems.
1.3 Purposes And Goals

The goal of this thesis work is to investigate the possibility of creating the call control parts of VM Call forwarding service, in an existing GSM call service environment. It also contributes to the existing body of knowledge of MTP Level3 User Adaptation Layer about how to integrate both pure SS7 and M3UA components.

The goal of the thesis is also to add a new feature to the existing GSM Call Service, supporting call forwarding, call deflection, transcoding, and external ISUP handling. This includes making the existing GSM Call Service to be more real world specific, where it can be applied as a testing framework.

The thesis work will be co-ordinated with another concurrent thesis work that aims to handle the media control part of the VM service, thus handling the media bearer differences(codecs and transmission rates), between TDM used as the bearer towards the VMS and RTP used as the bearer towards the GSM handset. The thesis work requires close co-operation with the system design at Mobile Arts.

The thesis serves particularly as a test framework, for testing the existing VMS application service in Mobile Arts Stockholm.
1.4 Thesis Structure

The rest of the thesis is organized in the following manner.

Chapter 2 describes GSM overview and the different protocols in the SS7 network.

Chapter 3 describes the call control use cases, the system capabilities for a call control, and the media control integration necessary for the system to function.

Chapter 4 describes the design of the GSM VMS system

Chapter 5 describes the system implementation

Chapter 6 discusses and concludes the thesis work

Chapter 7 describes the possible future work associated with this research
Chapter 2

GSM Overview and SS7 Protocols
The GSM overview is presented for readers to get a understanding of GSM systems their interworking and the different protocols involved. The SS7 overview is presented to describe in brief the different SS7 protocols and their interworking.

2.1 GSM Overview

GSM Network is a digital cellular network offering a number of services like call, SMS, fax, voicemail, and other supplementary services, with higher capacity and with optimal allocation of the radio spectrum.

The GSM network is structured into different entities like the Core Network (CN), and the Access Network (AN). The CN, handles the core switching services, and mobility management. The AN interfaces with the mobile stations and the CN.

The functionalities of GSM network is specified by the call and the location procedures that are available in the CN, which list and categorize the different types of calls and location based services that are available in GSM network.

GSM Network has different interfaces between its different elements. The interfaces are provided for facilitating information exchange, and for providing compatibility for different network elements from different manufacturers.

Figure 2.1 shows a GSM system composed of Mobile Stations (presented in 2.1.1), an Access Network (presented in 2.1.1), and a Core Network (presented in 2.1.2).
2.1.1 Access Network

The access network as the name states is for providing access to the mobile network. The access network handles the traffic and signalling between a mobile phone and Network Switching Subsystem (NSS)(also known as the CN). It also manages allocation of radio channels, paging, transmission, reception and transcoding of speech channels.

The access network which is also called the Base Station Subsystem (BSS) consists of Base Transceiver Station (BTS) and Base Station Controller (BSC).

The BSC controls a group of BTSes. The BTS is the base station which provides wireless communication between mobile phones and the core network. The BTS is placed in the center of the cell, and its transmitting power affects the size of the cell. The BTS provides encoding, multiplexing, modulating, and feeding the RF signals to the antenna. It also performs transcoding and rate adaptation.
The BSC manages the radio resources for the BTS, it is the link between the mobile station and the core network. The BSC handles the resources needed by assigning and releasing the frequencies that are required by a Mobile Station (MS).

The elements in the access network can be tracked to a Public Land Mobile Network (PLMN). A PLMN is a network identified by a Mobile Country Code (MCC) and a Mobile Network Code (MNC) which are managed and authorized by a managing entity with a purpose of providing telecommunication services. All the identifiable collection of devices and software that are used in the PLMN is categorized as the MS.

An MS consists of two important parts a Mobile Equipment (ME) and a Subscriber Identity Module (SIM). ME contains devices for transmission, reception, protocol handling and user interfaces. The SIM is a smart card containing the profile information of the user, user’s contacts, and other information. The SIM is provided to be detachable and can sometimes be reused with the handsets provided by different operators.

To provide anonymous(private) identification for subscribers in the GSM network, during call routing or roaming, GSM uses two unique identifiers. International Mobile Subscriber Identity (IMSI) is used as a 64 bit identifier, and is sent originally from the mobile phone to the GSM network. To increase security, and to prevent eavesdropping and tracking, Temporary Mobile Subscriber Identity (TMSI) is used as the identifier whenever an IMSI is required.

The subscriber’s mobile number is called as the Mobile Subscriber Integrated Services Digital Network Number (MSISDN). The MSISDN is the public identity of a subscriber, this is the number you get to share it with your contacts as your contact(mobile) number. The MSISDN with the IMSI are the two numbers used for identifying a subscriber in the GSM network.

### 2.1.2 Core Network

The core network performs call switching, routing and mobility management functions like roaming, and location based services. The essential elements of the core network are MSC, HLR, and VLR.

The MSC is a digital switching center conducting normal switch tasks and managing the network it performs the delivery service and is the main routing entity in the CN. Each MSC usually control several BSCs, it works through the interfaces
A and E, for setting up calls and managing subscribers. The MSC is in charge of all the signalling required for setting up, maintaining, tearing down connections in accordance with SS7 protocols.

The mobile originating call is handled by the Visiting Mobile-service Switching Centre (V-MSC), this is the landing module for the incoming calls. The calls which are towards the terminating subscriber(recipient) are handled by the Terminating Mobile-service Switching Centre (T-MSC). The gateway routing, call routing decision logic are handled by the Gateway Mobile-service Switching Centre (G-MSC).

To store all the subscriber information in the GSM network, service control points like the HLR are used. HLR is the central subscriber database storing all subscriber information, call forwarding information, service information, and subscriber location areas. HLR provides the subscriber information, and the required authorization checks during the setup of the call. The HLR uses the Mobile Application Part (MAP) protocol to exchange information with the communicating entities.

To reduce the load of the HLR and to help provide a location based storage, the VLR is used. VLR is used to lookup subscriber information, and call forwarding information. The VLR usually contains the subscribers associated with region the MSC serves. Subscribers move in and out of a region during roaming, the VLR updates the subscribers who roam into the region managed by the MSC in which the VLR is associated.

Call setup procedures in the GSM follow a set of protocols for how calls should be handled in a GSM network. Call Procedures are involved in Originating, Gateway and Terminating calls. The Originating call procedures are for handling calls that ‘originate’ from the network where the switch(MSC) is associated with. The Gateway call procedures are for handling calls passing through the gateway MSC, and the Terminating procedures for handling calls that terminate(or end), in the network associated with the switch(MSC).

In particular(for case in Figure 2.1), the V-MSC is responsible for the handling of call setup protocols like ISUP, BSS management protocols like BSSMAP, and direct transfer protocols like DTAP. The V-MSC receives the initial call setup request, with the various call setup parameters, the type of request, and the resources required. The V-MSC performs various subscriber and network status checks. The VLR which contains the details of the subscriber, decides whether the call can proceed. The V-MSC uses ISUP/IP signalling bearer to route the current call to G-MSC, through a transit module.
The G-MSC handles the gateway functions. It determines the routing and the transit logic of the call. It routes the incoming call through a transit module after the HLR routing checks. The HLR performs the required authorization checks, and tries to validate the call. The HLR also looks up the roaming number of the called party to complete the call.

To handle calls terminating (or incoming), the T-MSC module plays the role. The T-MSC notifies the subscriber of the incoming call, sets up the resources required for the call, handles allocation of radio resources required for the terminating subscriber to accept the completion of the incoming call [18].

**Transit Module**

The routing of the messages between the different MSC modules is done by the transit module. The transit module performs the number analysis, and determines the transit route of the call. The Transit module is also expected to handle ISUP/SS7 requests to route to external SS7 Networks [18].

**2.1.3 Location Procedures**

The location procedures module handle the registering and deregistering of subscribers to a network. The location information is stored in the VLR and the HLR. The VLR and HLR exchange information using the MAP protocol [18].

**2.1.4 Interfaces**

![Signalling protocol stack over A and E Interface](image)

Figure 2.2: Signalling protocol stack over A and E Interface
GSM provides the Abis interface for setting up and managing radio links. Radio Signalling Link (RSL) is used over the Abis interface between the BTS and the BSC. The interface located between the MSC and the BSS is called the A interface. The A interface supports multiple application parts like the Base Station System Management Application Sub-part (BSSMAP) and the Direct Transfer Application Part (DTAP).

The interface between the VLR and the MSC is called the B interface. All signalling with the VLR is done over the B interface using the MAP protocol [5] The VLR is interrogated by the MSC during the call setup and the location handling procedures. Supplementary services are also handled and setup through the B interface. [3, 4]

The interface located between the HLR and the G-MSC is called the C interface. The routing information, and the location updates are transferred through the C interface during the call setup. The interface located between the VLR and the HLR is called the D interface, there is a exchange of location and subscriber management information using the D interface.

The MSC communicates with another MSC using the E interface. The ISUP protocol is used to communicate over the E interface.
2.2 Signalling System 7 Protocols Overview

SS7 is a set of telephony protocols which are used in the majority of the PSTN networks to setup, teardown telephone calls. The SS7 protocol stack for cellular-specific services consists of Signalling Connection Control Part (SCCP), Message Transfer Parts, and also a Stream Control Transmission Protocol (SCTP) transport mechanism. The entire suite of protocols associated with SS7 is called Signalling Transport (SIGTRAN).

Message Transfer Part (MTP) is used for reliable communication in the SS7. It also takes care of messages being non-duplicated and delivered in sequence between the different communication entities in the SS7.

MTP has three levels, the lowest level defines the characteristics of the digital signalling link, with physical interfaces (E1, T1, DS-1, V.35). MTP Level 2 provides end to end delivery with features like flow control, and validation of
message sequences. MTP Level 3 provides message routing between the various signalling points in the SS7 network. MTP uses a routing label for message routing, the routing labels are Originating Point Code (OPC), and the Destination Point Code (DPC).

SCCP is used to provide both connection oriented and connectionless network services in an SS7 network. SCCP also acts like a transport layer for Transmission Capability Application Part (TCAP) services. All the non-circuit related information that needed to be exchanged are done using the TCAP protocol. This happens between the various signalling points in the SS7 network. TCAP is delivered using SCCP connectionless service. TCAP messages are contained within the SCCP portion of a Message Signalling Unit (MSU). A TCAP message is comprised of a transaction portion and a component portion.
2.2.1 M3UA

MTP Level3 User Adaptation Layer aims to support the transport of SS7 user signalling traffic, over IP using SCTP.

This allows the network to provide services which are more flexible and efficient through IP.

GSM call service is modeled to support ISUP and SCCP over IP. GSM call service provides support for DTAP, BSSMAP, ISUP, over IP.

2.2.2 ISDN User Part

ISUP defines the protocol and procedures used to set up, maintain and clear calls. ISUP call setup deals with the control signals (and parameters) that are required when first initiating the connection for a call. The parameters contain all the information that are required for a call setup. ISUP call proceed indication are the signals (and parameters) sent when a call is accepted. ISUP conveys the circuit information used in a call using a parameter called Circuit Identification Code (CIC).

ISUP/SS7 is ISUP protocol run over SS7 protocols using MTP for transmission, ISUP/IP is ISUP protocol run over IP using SCTP for transmission as enabled by M3UA [14]. Figure 2.3 shows ISUP/SS7 and ISUP/IP protocol stack.

Call Setup

GSM Call Service uses ISUP/IP as signalling, ISUP/IP is described by Erlang messaging and data structures, sent through the M3UA layer.

ISUP serves for carrying the signalling information like the CIC assigned, call forwarding information, calling information, redirection information, and also the cause.

ISUP initial message describing the setup of the call is specified by the ISUP Initial Address Message (IAM), the ISUP parameters necessary for setting up a normal call are

- Message Type
- Called Party Number
• Calling Party
• CIC

The parameters for call forwarding are

• Redirecting Information
• Called Number
• Calling Number
• Forwarding Number
• Redirection count
• Generic Number
• Original Called Number
• Cause Indicator

Redirection number contains the forwarding subscriber’s MSISDN, the called number is the MSISDN where it is being forwarded to, the redirection count has the number of times a particular call is being forwarded.

The ISUP access handler maps the ISUP IAM message to the ISUP setup request and sends to the VMS handler through the SS7 stack.

Call Proceed Indication

In the GSM Call Service, the call proceed is notified by two ISUP messages [6],

• ISUP ACM
• ISUP CPG

ISUP ACM is the address completion message sent by the GSM call service to both the originating and the termination BSC’s to setup the alerting(ringing) in the mobile subscriber’s phone.

ISUP CPG Is the call progress message.
CIC Allocation

The allocation of CIC to individual circuits is determined by bilateral agreement in accordance with applicable predetermined rules [15].

The 5 least significant bits of a CIC represents the timeslot assigned to the communication path, and the four spare bits are for CIC extension.

Call Confirm

In the *GSM call service*, the call confirm is notified by ISUP ANM Message, which is the ISUP answer message.

This is the phase where the alerting stops, indicating that the terminating subscriber has answered the call, this is also the phase where the media is setup between the calling and the called party [6].

In the *VMS*, the call confirmation is notified by ISUP setup confirm message. This is when the VMS starts playing the *welcome* prompt, and starts recording the voice mail.

Call Release

The call is released using ISUP release and ISUP release complete, ISUP release contains the cause, which indicates the reason for the release.

2.2.3 Mobile Application Part (MAP)

Communication between the G-MSC and the HLR, and between the VLR and the HLR is done based on the MAP protocol.

The HLR is a Service Control Point (SCP) in the SS7 network, configured to handle MAP messages from the VLR and the G-MSC.

In the B interface, the MAP/IP is used, which needs to be mapped to MAP/SS7. MAP messages are used for supplementary service handling, for setting up the subscriber supplementary services configuration like call forwarding handling, activating, deactivating, provisioning and erasing basic services for a subscriber.

MAP messages also play a role in Routing, Location handling, Retrieving information for handling incoming and outgoing calls, the information specifies how
the call should be handled in exceptional cases.

Information for incoming and outgoing calls, are sent between the G-MSC and the VLR, subscriber location information are sent between the VLR and the HLR, routing information are sent between the G-MSC and the HLR.

SCP entities like the HLR and the VLR also decide on the call forwarding logic, like early call forwarding by the HLR, subscriber absent, subscriber busy, or network congestion call forwarding by the VLR.

MAP in the SS7 protocol is carried over TCAP, the MAP parameters are mapped to the TCAP parameters.

Since the design of the system is to handle MAP/IP in the B interface, and MAP/SS7 in the C and D interfaces, there are MAP handlers to handle the bearer differences.

The transactions in a TCAP protocol are mapped to the transactions in the MAP protocol, The MAP handlers translate the OPEN and the CLOSE MAP requests to the BEGIN and END TCAP requests. Refer 4.2.3 for more MAP handling.
2.3 GSM Call Service

GSM call service developed during the course Project CS 2011[19], will be enhanced and integrated with Mobile Arts (MA) VMS.[18].

GSM call service provides an MSC switch functionality for voice calls with RTP as media bearer, and ISUP over IP as signalling bearer.

GSM call service does not support supplementary service functionalities, bearer capability for European-carrier 1 (E1)/TDM links, and SS7 signalling messages like (ISUP, MAP, TCAP), which are required for VMS control over E1/TDM links.

Note that ISUP in GSM Call Service is provided through Erlang messaging only.
Chapter 3

GSM Voice Mail Service and its Call Control
VMS is a service that allows receipt and storage of voice messages. VMS can be used in the GSM network to provide subscribers with automatic answering features when a subscriber cannot be reached for a variety of reasons. Interactions with the VMS can be realized by the supplementary services provided by GSM. The supplementary services provided by GSM include call forwarding and call deflection [12, 10].

Call control requirements can be realized from two different perspectives, subscriber use case perspective and from a system requirements perspective. The user perspective describes in detail the call control that is required for different call forwarding use cases like forward calls all the time, forward calls only when user is busy, etc; and for subscribers to handle subscriptions like registering to a particular supplementary service, unregistering to a supplementary service.

The system perspective describes the call control functionality required in GSM components to handle call forwarding messages, and the handling of supplementary services.

MSC modules (V-MSC, G-MSC, T-MSC), require support to handle call forwarding messages (DTMF signalling, DTAP, etc.).

Service control points like HLR and VLR need to handle the different subscriber authorization for call forwarding, call forwarding decisions like when to forward a call. To test if the system has capabilities to handle call forwarding, and whether the call has reached the maximum number of forwarded hops.

Call control finally addresses how call control parameters include information for the circuit that is setup to support the voice (media) flow between the various call hops from one signalling point to another.

Call control contains all the information about how a particular call circuit is setup, for GSM components that transmit the voice (media) through the circuits.

### 3.1 Use Cases

The following are the call control features of VM from a mobile subscriber perspective,

**Call Forwarding Unconditional**

The served mobile subscriber can use the Call Forwarding Unconditional
(CFU) supplementary service to forward all incoming calls for the specified basic service(s) [11, 12].

Call Forwarding Busy
The served mobile subscriber can use the Call Forwarding Busy (CFB) supplementary service, to forward incoming calls if the mobile is determined Network Determined User Busy (NDUB) or User Determined User Busy (UDUB) for the specified basic service(s). [11, 12]

Call Forwarding On No Reply
The served mobile subscriber can use the Call Forwarding No Reply (CFNRy) supplementary service, to forward incoming calls if the call is not answered within the period defined by the no reply condition timer for the specified basic service(s). [11]

Call Forwarding On Not Reachable
The served mobile subscriber can use the Call Forwarding Not Reachable (CFNRc) supplementary service to forward incoming calls if the mobile subscriber is not reachable. [11]

Call Forwarding Multiple Hops
The system may support scenarios where call forwarding happens between multiple subscribers defined in multiple number of hops.

VMS Services
GSM Call Service VMS integration should support deposit and retrieval of voice mail messages

Dual Tone Multi-frequency (DTMF) services should be supported to enable VMS communication

DTMF services may support configuring VMS subscriber options like subscriber type, service keys, white listing, black listing, subscriber information, fax number, greeting messages, avatar ..etc

DTMF services may support configuring VMS subscriber profile options of setting Electronic Mail (E-Mail), Multimedia Messaging Service (MMS), SMS, and fax notification options.
Call Forwarding Handling [12]

Registration
Subscriber can register the forwarded to number, and the information as to whether all calls or all calls of a specific basic service should be forwarded.

Erasure
Erase a previous registration

- The subscriber can specifically erase a previous registration with an appropriate control procedure.
- The subscriber can register information for a particular supplementary service using another directory number.
- The network administrator can withdraw a supplementary service.

Activation
Subscriber can be allowed activation to all subscribed basic service groups against which a supplementary service code is registered. Service shall be subscribed and registered.

Deactivation
Subscriber can deactivate to a previous activation, and register information for a Supplementary Service (SS)-code by overriding, or deactivate by withdrawal.

Interrogation
Subscriber can obtain information about the data stored in the PLMN, information such as SS-status, the associated forwarded-to number.
3.2 System Capabilities

The following are the call control features from a system perspective.

**V-MSC Features**
V-MSC should be able to handle DTAP Call Forwarding *handling* messages and map it to corresponding MAP messages in the VLR for CFU, CFNRc, CFNRy, and CFB.

The V-MSC shall also provide support for DTMF signalling messages, CIC allocation and deallocation.

**G-MSC Features**
The G-MSC should be able to provide an external MAP handler for map incoming and map outgoing requests. This is where the signalling bearer differences between MAP/SS7 and MAP/IP are handled.

G-MSC should also provide support for call forwarding notification messages.

**Terminating Mobile-service Switching Centre Features**
The T-MSC should provide support for handling external ISUP through ISUP access handler, and DTMF signalling support.

**ISUP Access Handler**
ISUP access handler should be able to use the SS7 Tieto stack for ISUP/SS7 handling, and should handle CIC provisioning.

It also should provide means to convert ISUP/IP to ISUP/SS7 and vice versa. The signalling bearer differences between ISUP/IP and ISUP/SS7 are handled by this module.

**VLR Requirements**
VLR should handle CFB using the Mobile Additional Function MAF008 [7] authorizations examination, where authorization routines for CFB is specified.

The VLR should handle CFNRy using Mobile Additional Function MAF009[7] authorizations examination, where authorization routines for CFNRy is specified.
VLR should handle CFNRc using Mobile Additional Function MAF010 [7] which is the examination of CFNRc when either the mobile subscriber is not reachable in the VLR, or when there is no paging response or when there is a radio congestion.

**Home Location Register Requirements**

HLR should handle the following call forwarding decision cases: CFU, and CFNRc (when the mobile subscriber is deregistered or purged in the HLR or not reachable in the VLR). HLR should also implement the mobile additional functions MAF007[7], MAF010(subscriber purged from HLR).

HLR should support the registration, erasure, activation, deactivation, and interrogation of the call forwarding services.

HLR should store information related to call forwarding. The basic service group code, forwarded-to number, notifications to the subscriber, and the value of the *SS-code* such as the Provisioning, Registration, Activation, and the HLR induction states.
3.3 Media Control Integration

Signalling of call control entities is not complete without setting up the circuit for the voice to flow. The circuit information is found in the CIC parameter, carried in the ISUP messages.

In the ISUP/IP part of the network, the CIC information contains 32 available timeslots, for media, the timeslots are mapped to the 5 least significant bits of the 32-bit CIC [15, 16].

In the ISUP/SS7 part of the network, the CIC information directly translates to a Generic Telecom Hardware (GTH) hardware information. The GTH hardware is identified by a GTH head, GTH span, GTH timeslot.

The media bearer difference handling between RTP with Enhanced Full Rate (EFR) codec and GTH with A-law codec is provided by concurrent research work which aims to provide the media control for GSM VMS.

The circuit setup starts with each MSC worker seizing a switch from the connection service, the connection service provides virtual switching, the mobile originating and terminating worker seizes an access switch, while the gateway worker seizes a service switch.

The connection service creates a context when the switches are seized in an order, once a context is created, the media path is setup, at the receipt of the ANM message or the ISUP setup confirm message. The RTP and the GTH termination agents, manage the termination differences respectively in the Media Gateway (MG). [18]

The role of the call control is complete, with the seizing of virtual switches in the connection server, The connection server talks with the MG in the MSC and the VMS using the megaco protocol[17], to signal the termination additions and updates.
Chapter 4

Design
4.1 Architecture

![System Architecture Diagram]

Figure 4.1: System Architecture
The Mobile User Service (MUS) module in the Mobile Switching Centre handles the mobile user services like mobile originating calls, mobile terminating calls, and routing calls through the gateway module.

The application platform in the MSC consists of a connection service, which provides virtual connections (switching), for MUS modules that require access to the termination. The MUS modules use the connection service to create the connection setup for a call. The connection service instructs the Media Gateway Controller (MGC) to prepare the MG to setup and teardown terminations required for a call.

Termination is a source or a sink for one or more stream of information [13], and a context is formed by an association between a collection of terminations. A termination is required for abstracting out the information about a information stream. For example a termination could either represent a RTP traffic (having audio with codec EFR), or traffic from E1/TDM (having audio with codec A-law).

The communication between MG and MGC is done using the megaco protocol [13]. The MG has termination access agents, which manage the terminations, and handles the megaco termination requests, like add, subtract, and modify.

The add request adds a termination (say T1), by adding a RTP termination on the BSC side, and an another subsequent add request will add a E1/TDM termination (say T2) on the MSC side, now the two terminations form a context. The subtract request removes a termination from a associated context. The termination is removed when a circuit is no longer required. The modify request modifies the termination properties, to include more stream source/sink, or to remove existing ones. thus the connection service abstracts out these information, providing only the details of the circuit to the MUS modules.

The external HLR is provided in the SS7 network, which stores the subscriber information, while the VLR is a part of the MSC node. The Transit Module (TM) handles the transit of all the messages in the MSC to each of the worker process.

Introducing E1/TDM access agents to work alongside RTP access allows the VMS, BSC, and the regular non Call Forwarding (CF) calls to work without any change. Having a transcoding client eliminates the need of transcoding when the codecs from BSC are the same, with respect to non CF calls.

The ISUP access handler, provides external ISUP access towards the VMS. The
VMS uses a GTH hardware, which are connected to the telephone network via E1 or T1 links (2 or 1.5Mbit/s) and are controlled over ethernet.

Having an External ISUP agent allows a modular design, and to connect to the SS7 network only when required for external calls, this is helpful for non SS7 traffic. The ISUP access handler, is designed to work like the rest of the call workers, like mobile originating, mobile terminating, and gateway.

The transit module is extended to handle both the SS7 and non SS7 traffic, the traffic to the VMS is designated to route towards the external ISUP, instead of the terminating worker. This allows the VMS call to follow call control sequences that is similar to a regular call, but differ only with regard to termination. The calls to VMS terminate in the isup access handler, and is sent through the SS7 network.

The architecture follows a mix of the M3UA layer, and the SS7 network, with components to handle differences and provide compatibility. The architecture also borrows a great deal of the design from how the GSM call service is designed, and is extended to provide compatibility.
4.2 Components

Since the implementation will try to enhance the already existing GSM Call Service, the design focuses on what new components will be added to GSM Call Service and what existing components will be changed.

MUS will be required to handle call forwarding and call handling traffic using co-ordinator processes, so that it doesn’t break the existing design. The co-ordinator process will be required to handle all supplementary services provided by GSM, which can be enhanced in future if it is required for GSM VMS to support more supplementary services.

The VLR component is required to handle MAP requests, from the HLR, and MSC so there is a need for map handlers, GSM Call Service does not provide map handlers, but only simulates map messaging. The VLR component is also required to handle authorizations for subscription checks, and call forwarding checks.

MAP handlers will be used by three components, namely the VLR, the MSC and the HLR to communicate with one another. The design should focus on having a generic map handler that can also serve component specific custom messages if required.

Finally the TM requires external ISUP handling, here external refers to the ISUP messages that uses MTP for transmission in the SS7 network. This is enhancing GSM Call Service to handle ISUP/SS7 traffic, and integrating it in the SS7 network.
4.2.1 Mobile User Service

Figure 4.2: Mobility User Service components
The MUS has modules which handle the call originating, gateway, and terminating functionalities of a call. This will suffice for a regular call where the scenario for forwarding never happens. However to support VMS operations the design has to be extended to handle cases when a call can be forwarded.

Imagine a scenario where a terminating call is not completed either because the user is busy or there is network congestion. In either case the MUS simply doesn’t have the functionality to forward the call to another subscriber.

Support for call forwarding can be added without much effort, since the *GSM Call Service* already provides a routing abstraction through the TM. The functionality can be added in the TM to route forwarded messages, and to keep track of the number of hops.

This design will suffice if the traffic is always in the area controlled by the MSC, but real world scenarios require traffic to be routed to external gateways and network. Hence the TM should support external ISUP messages handling, and transit. This is realized through the external ISUP handler in the TM. Detailed information about the TM will be covered in section 4.2.4.

There is also a need for taking user preferences into account, when providing supplementary services. The mobile subscribers(or users) require handling of supplementary services, like registering, unregistering. A co-ordinator process is added in the MUS to provide handling of supplementary services. A concurrent thesis work uses the co-ordinator process to handle supplementary services like Unstructured Supplementary Service Data (USSD).

The supplementary services handling messages are sent as DTAP messages. *registerSS, eraseSS, activateSS, deactivateSS, interrogateSS* for registering, erasing, activating, deactivating, and interrogating the supplementary services.

The behaviour of the co-ordinator process for every message is specified here.

- Register_SS_MSC message is received as a DTAP message, and sent as a MAP message to the VLR without checking the contents of the service to be registered to.
- Erase_SS_MSC message is received as a DTAP message, and sent as a MAP_ERASE_SS message to the VLR.
- Activate_SS_MSC message is received as a DTAP message, and sent as a MAP_Activate_SS message to the VLR.
• Deactivate_SS_MSC message is received as a DTAP message, and sent as a MAP_Deactivate_SS message to the VLR.

• Interrogate_SS_MSC
  Transfer the message to the VLR with MAP_Interrogate_SS request without checking the contents.

4.2.2 VLR

The VLR in the GSM Call Service does not handle MAP messages. The B interface (between VLR and MSC), does not require MAP messages, as they are closely integrated, and communication can be done using language specific message passing.

The call forwarding authorizations for VLR fall under different categories like NDUB, UDUB, CFNRc, and CFNRy.

NDUB is when the VLR determines that the user is busy, and cannot be reached, this happens when the PAGE request sent by the VLR to contact the mobile subscriber fails.

UDUB is when the VLR determines that the user is busy, and cannot be reached, this happens when the user receives the call, and decides to reject it.

CFNRc happens when there is congestion, or when the subscriber has detached from the VLR and CFNRy is when the subscriber never answers the call.

For each of these scenarios, the VLR checks whether a CF option has been subscribed by the user and is enabled by the operator, the VLR then performs a series of tasks to enable the ISUP message to carry sufficient CF information.

The Mobile Additional Function (MAF) are a series of functional checks performed by the VLR (and the HLR), to determine whether a call can be forwarded. If the check is successful the VLR populates the forwarding number, the forwarded-to number, the forwarding hop count, and routes the message.

The VLR provides a MAP handler to communicate with the external HLR, and to provide map handling. Every subscriber handling request is handled in the VLR to handle a request to register, erase, activate, deactivate, and interrogate [7, p. 617].

For the supplementary services the VLR shall store the service state information received from the HLR. This is the SS status for every subscriber.
• Provisioning State
  Whether the particular SS has been provisioned by the network operator.

• Registration State
  Whether the user has registered to the particular SS.

• Activation State
  Whether the particular SS has been activated.

4.2.3 MAP Handlers

MAP handlers are provided for use by VLR, and MUS to establish MAP communication with HLR. MAP uses the TCAP protocol and is designed to have sessions very similar to a TCAP session.

A MAP session starts with the OPEN request, and ends with the CLOSE request, while a TCAP session starts with a BEGIN and ends with END request [8, 7].

To separate connections between originating and termination requests, the design emphasizes, a MAP out and a MAP in handler. The MAP out handler is for any outgoing map session, initiated by a map user. MAP user is one of VLR, HLR,
or the MUS. The MAP in handler is for any incoming map session accepted by a map user. The MAP message that operate over the B, C, and D interfaces can be classified into one of the categories, as either a MAP in or a MAP out handler. This greatly simplifies the design for handling different MAP messages.

4.2.4 Transit Module

VMS requires the use of tones and announcements, to communicate with the calling subscriber about the status of the call, and the necessary actions that are required to complete the voice mail.

TM is enhanced to send tones, and announcements, in case of busy, congestion, alerting etc. This is done by having a mapping between the ISUP cause(codes), and the various tones that need to be announced . Tones also include special information tones due to numbering plan error, congestion tone, and other custom tones in the future.

TM sends tone using the connection service, the TM seizes a virtual connection(as a service user) to send custom tones when a call is ongoing. The message is sent with parameters specifying the virtual switch, the virtual port, and the tone id.

TM also handles the routing of external ISUP to the external ISUP handler. The external ISUP handler is managed by the TM.

The design focuses on having the external ISUP handler communicate with the Tieto SS7 stack, to receive the ISUP/SS7 messages.

4.2.5 Connection Service

The Connection Service (CS) should be enhanced to provide support for tones(announcements and DTMF). The MUS worker or the TM specifies the termination id, the virtual switch id, and the virtual port.

The CS, instructs the megaco client to send a megaco modify request for DTMF and tone sending.
4.3 Call Control Interfaces

*GSM Call Service* interfaces are to be extended to provide support for VMS service. Interfaces highlight the roles played by the components to interwork with each other. The design has to consider a lot of protocol differences and provide abstraction over protocols by using handlers.

MAP handlers used by VLR, HLR, and MUS provide interfaces that allow the map messages to be sent over IP or SS7 (MAP over IP, MAP over SS7).

This helps the existing *GSM Call Service* to integrate easily with SS7 traffic. The following section covers the call control interfaces between the various components, the map handler interfaces for handling TCAP messages from a TCAP user, the connection service interface, and the ISUP access handler interface.

4.3.1 Map Handler Signals

Map handlers are used by VLR, HLR and the MUS. The MAP handler handle TCAP messages from the Tieto SS7 stack, and translates them into MAP messages.

The TCAP session starts with a bind request, from the SS7 stack, the *BIND* request binds the map user to the TCAP session; to close the session with the TCAP user the *UNBIND* request is used.

Once the TCAP user is bound to the TCAP service in the Tieto SS7 stack, the session is intitated by the *BEGIN* request. The MAP handler, translates the *BEGIN* request to a *MAP OPEN* request.

To invoke any MAP services, the *TCAP INVOKE* request is used, providing the service as the parameter name, this is translated to the *MAP INVOKE*.

All the service specific request are sent as parameters, which are directly mapped to the MAP parameters.

This is the first role of the MAP handler, the next operation is to convert (if necessary) the MAP message into the message format required by *GSM Call Service* components if the MAP user is in the B interface (VLR and MUS).
4.3.2 VLR Interface

The VLR uses the map handler to initiate any outgoing map session with the HLR or the MUS. The MAP session with the HLR is sent as MAP over SS7[7, p. 272], and the map session with the MUS is sent as MAP over IP.

4.3.3 HLR Interface

_GSM Call Service_ uses a HLR as a MUS component, the _GSM VMS_ will use the HLR provided by Mobile Arts.

This HLR is an external HLR containing the data of the subscribers of Mobile Arts; interfacing with the real world HLR would require the use of MAP handlers.

The MUS and the VLR uses the map handler to communicate with the HLR, this follows the same pattern as specified in the section 4.3.1.

4.3.4 MUS Interface

MUS interface with VLR and TM will be re-used from _GSM call service_, the interfacing with HLR will be enhanced with MAP handlers as specified in the section 4.3.1.

4.3.5 Connection Server Interface

Connection service in the _GSM Call Service_ will be enhanced to provide services to the MUS modules to send tones and announcements. The MUS modules request a _service_ user privilege from the connection service, to send the DTMF tones and announcements.

4.3.6 ISUP Access Handler ↔ Tieto ISUP stack

Tieto SS7 signalling stack is used by Mobile Arts for SS7 communication, the interface between ISUP access handler and Tieto ISUP stack is to setup, maintain, and teardown, ISUP connections as required.

ISUP connection is setup by binding to the Tieto stack, using the _ISUP BIND_ message, _ISUP IAM_ is received by the ISUP access handler when there is an incoming call. The _ISUP IAM_ is sent as _ISUP SETUP REQ_ to the Tieto Stack, with important parameters like calling party number, called party number, and
the redirecting number.

The termination side, confirms the call using *ISUP PROCEED* request, this is usually a VMS which sends the message if the call has been forwarded; with the receipt of the proceed message, the connection is established. ISUP also allows circuits to be reserved and unreserved.

To teardown a ISUP connection, the ISUP access handler sends the *ISUP RELEASE* with the corresponding release code, the release codes are useful in announcements and DTMF signals.

### 4.3.7 Transit Module ↔ ISUP access handler

The transit module interfaces with the isup access handler to start and stop it. It also converts certain ISUP parameters like ISUP CIC, ISUP calling party address, and ISUP called party address used internally in *GSM Call Service*, to the format required by ISUP access handler.
4.4 Handling Bearer Differences

The real challenge in getting the system to work, in other words to send a M3UA traffic to the SS7 network and back, is to handle the difference in the protocol.

4.4.1 ISUP Mapping

ISUP access handler provides ISUP CIC handling. It also maps the primitive ISUP message types to ISUP setup requests that should be sent to the Tieto SS7/ISUP stack. The following table specifies the ISUP mapping.

**Mapping**

**Call Setup Stage**

| ISUP IAM       | → isup_setup_req |
| ISUP ACM       | → isup_setup_req |
| ISUP CPG       | ← isup_proceed_req |
| ISUP ANM       | ← isup_setup_resp |

**Call Connected Stage**

| ISUP CPG       | → isup_progress_ind |
| ISUP CPG       | ← isup_progress_req |

**Release Initiated By MSC**

| ISUP REL       | → isup_release_ind |
| ISUP RLC       | ← isup_release_resp |

**Release Initiated By VMS**

| ISUP REL       | ← isup_release_req |
| ISUP RLC       | → isup_release_conf |

CIC is chosen and sent in the ISUP IAM message to the ISUP access handler, the *isup_access_handler* maps the ISUP IAM request to ISUP setup req, where the ISUP head contains the timeslot and the span taken from the CIC.

The VMS deposit or retrieve action is determined if the roaming number is present. If the roaming number is absent it corresponds to a retrieve action, if the roaming number is present it is deposit.
ISUP deals with the following parameters affecting call forwarding

**Multiple Call forwarding and ISUP parameters**

**Calling Information**

- Hop1=(cdpn=B, redirectNumb=_, origNumb=_)
- Hop2=(cdpn=C, redirectNumb=B, origNumb=B)
- Hop3=(cdpn=D, redirectNumb=C, origNumb=B)
- Hop4=(cdpn=E, redirectNumb=D, origNumb=B)

**Redirecting Information**

- Hop1=(redirectCounter=0, redirectInd=, redirectReason=)
- Hop2=(redirectCounter=1, redirectInd=(Call Diversion), redirectReason=(CFU, CFB,))
- Hop3=(redirectCounter=2, redirectInd=(Call Diversion), redirectReason=(CFU, CFB,))
- Hop4=(redirectCounter=3, redirectInd=(Call Diversion), redirectReason=(CFU, CFB,))

ISUP setup request parameters contains ISUP head which has the unit, span, timeslot information.

The present design of the *GSM Call Service* is that the operator associates the CIC and termination IDs in the RTP termination agent and the E1/TDM termination agent.

When new terminations are added, the MG notifies the MGC, the MGC updates the termination agents (both RTP, and E1/TDM). The operator can then associate a CIC to a termination through the agents (RTP and E1/TDM).

The MG only knows the termination id. The termination ID is used to assign switches in the CS. The CIC is used for circuit allocation in the BSC and VMS.

The MUS requests the CIC and Termination pair from the RTP and E1/TDM agents, and releases them after call completion. [18, p. 37]
Chapter 5

Implementation And Test
The following chapter describes the implementation methodology, the environment setup, and the different levels of implementation groups; with each group listing the different levels of required effort. The implementation also gives a big picture of the design objectives in every phase and their expected outcome.

5.1 Methodology

The methodology adopted for implementation was a customized mix of Scrum and Waterfall. Scrum is an agile software development which is interactive and incremental, while Waterfall model, is a sequential design process, starting from requirements, design, implementation, followed by maintenance.

The requirement gathering phase for the thesis required the Waterfall model to understand the standards and specification, and listing out the important requirements for the thesis. The Scrum model was customized to be more like an evolutionary model, to fit a team size of two.

The Scrum model was more useful during the implementation, to release incremental working features, and to get quick feedback from the company.

5.2 Environment Setup

The environment consisted of BSC, MSC, HLR, VMS, and Tieto SS7 nodes; The VMS required setup and configuration to use the tones and required services.

Use of nanoBTS requires a license requirement, and the environment setup required some planning to install the GSM Call Service, in the SS7 network.

The challenge was in enabling VMS to bind with the SS7 stack to use the ISUP service, and binding the MUS module of the GSM Call Service to use the ISUP, and MAP services provided by Tieto ISUP stack.

The hardest problem was in configuring the Tieto SS7 stack, as it required detailed understanding of SS7, and SIGTRAN gateways. Since this was done in close cooperation with Mobile Arts, we received assistance in setting up the Tieto SS7 stack as required.
5.3 Implementation Groups

The implementation was based on the existing GSM Call Service design. To enhance the already available GSM Call Service, by categorising the requirements into three groups.

The first group were the modules that were present in GSM Call Service, which could be reused as-is.

- **TCP/IP Server**
  The module receives the DTAP and BSSMAP messages from the BSC sent over a TCP connection. It could be reused as-is, as the GSM VMS used the same design structure as followed by the GSM Call Service to connect the AN to the CN.

- **VLR call handler**
  VLR Incoming Call Handler (VLR ICH), is responsible for VLR decisioning for incoming calls, and VLR Outgoing Call Handler (VLR OCH) is responsible for VLR decisioning for outgoing calls. Since the GSM VMS is required to support regular calls and there won’t be any change in the way a non CF call will be supported, these modules can exist as-is.

- **Tieto SS7**
  The Tieto SS7 stack provided by Mobile Arts will be reused as-is without any changes.

The second group were the modules that were present in the GSM Call Service, which could be used with some enhancements.

- **MUS**
  The MUS required changes to call originating, call terminating and the gateway modules to handle support for call forwarding scenarios.

- **TM**
  The transit module, required changes to handle ISUP access handler, and external ISUP routing.

- **VLR**
  The VLR required changes to handle map handlers, and the VLR decisioning cases used for call forwarding.

The third group were the list of modules that were to be newly added
MSC Co-ordinator
Since GSM Call Service did not provide support supplementary services, there was no requirement for a co-ordinator process. In GSM VMS a co-ordinator process is required to support, supplementary services provisioning and handling.

Map handler
MAP handler is a new module, which can provide protocol abstraction over GSM Call Service map handling modules and SS7 map users (like VLR, HLR) that deal with MAP traffic.

5.4 Design Phases
The first design phase consisted of designing and implementing the requirements, in a series of iterations. The first phase was planned to setup the existing system and run it in the SS7 environment. While the second phase consisted of writing the ISUP call control parts, to integrate it with the VMS. Once the integration was done the third phase was to integrate the call control with the media control provided by the concurrent thesis.

Phase 1
The objective of Phase 1 was to get the existing system(GSM Call Service) integrated and running with the Mobile Arts environment. This included implementing ISUP handlers, and co-ordinator processes to communicate with SS7 protocols. The objective was also to enhance the existing GSM Call Service to function with call forwarding scenarios.

The implementation goal was focussed on keeping the CF scenarios easy to test, and avoided complex multiple call forwarding hops; since the original goal of the thesis was to try out the possibilities of integration of ISUP over IP and ISUP over SS7 (including MAP over IP and MAP over SS7).

The implementation goal also focussed on keeping the supervision structure provided by Erlang/OTP simple, and easy to test. Stability of the system, and requirement of different tones and announcements for the VMS were moderately prioritized.

The output of the Phase 1 was a system, where the MUS modules, can send tones, and announcements by using the CS as a service user.
Phase 2
The objective of Phase 2 was to implement the ISUP access handler, which can be controlled by the TM, the objective also was to enhance TM to handle external ISUP. The final goal was to send a ISUP message from GSM Call Service across the SS7 network (to bind and unbind) and back.

The output of Phase 2 was enabling ISUP communication, with external ISUP handling and routing by the TM.

Phase 3
The Mobile Equipment sends and receives audio with the codec EFR, and VMS sends and receives audio with codec A-law. The concurrent thesis, dealt with transcoding, and rate adaptation between the two codecs. Phase 3 of the project dealt with early integration of GSM Call Service with external ISUP handling, and the transcoder.

The output of Phase 3 was a partially functioning GSM VMS, capable of supporting non concurrent calls for deposit and retrieval of voices messages from the VMS.

Phase 4
Phase 4 dealt with making the system more useable for testing the VMS service. It also dealt with adding DTMF tones and announcements to enable subscribers to select their services. This was partially completed, as the project goal was already realized with the completion of deposit and withdrawal of messages from the VMS.

5.5 Testing
The original testing objective was to reuse the test cases that came along with GSM Call Service, it included complete system and unit-testing using Erlang rebar.

As the thesis objective was not to deliver a stable working system, system tests were not prioritized. Testing was done to verify call control use cases, which included CFU, CFB, CFNRc, CFNRy.

CFU and CFB were verified by having a subscriber’s list, whose IMSI was configured in the HLR to always forward to the subscriber’s voicemail. This was done
by manual testing and didn’t require any test cases.

CFNRy and CFNRc were tested, by simulating network congestions and subscriber absence, enabling the call to reach the subscriber’s voice mailbox, when call forwarding was enabled.

Main focus of testing were the integration of call control and media control modules. The following were covered during integration testing.

- CS
  Since the MUS modules maintain states for every call, the connection to CS should be stable and provide correct virtual switching and virtual ports access (for circuit switching) for the MUS modules, when a call is forwarded.

- CIC transmission
  Since the CIC conveyed the circuits used in a call, it was to be effectively conveyed to the media control modules. The terminations used in the access agents, should have a CIC mapping, the tests were to ensure, the mapping was proper, and each call had a valid CIC.
Chapter 6

Discussions & Conclusions
Traditional telephony networks, have the advantage of being extremely reliable, but also expensive. The converge networks (SS7 over IP) specified by the SIGTRAN family of protocols (SCTP, M3UA) brings with it the advantages of decreased cost of infrastructure, and improved ease of management. The following are the highlights of the advantages provided by using convergence networks.

6.1 Features

SS7 messages transported over IP networks, adopts common transport protocols, and adaptation modules to provides features like flow control, identification of voice circuits, error detection, retransmission, security, and congestion avoidance etc. Size restrictions posed in the SS7 network, are no longer a problem in the IP networks. VMS TDM Call Control provides these features by the use of SIGTRAN(M3UA) protocols.

6.2 Performance

The standard (ITU) specifies the performance requirements, and response times of the application. For example ITU specifies the end-to-end call setup delay cannot exceed 10 to 20 seconds after the initial ISUP IAM. VMS TDM Call Control, is designed to satisfy user expectations and ITU standards for performance.

6.3 Security

SS7 messages transported over a private internet are relatively safe, and security measures can be deemed as necessary by the operator, but for SS7 over IP transported over public networks, the use of security measures is mandatory. Security over IP is provided by the use of IPSEC, which provides authentication, integrity, confidentiality, and availability. VMS TDM Call Control uses SIGTRAN family of protocols (SCTP, M2UA, M3UA), and hence meets high security expectations.

6.4 Cost

The standardization forum 3gpp has specified GSM, while ITU-T specifies ISUP call control protocol, relying on the expensive telecom specific TDM. The research work finds its areas where cheaper alternatives are required for expensive telecom hardwares. It implicitly also provides the GSM call service which can be used in a academic environment, with a nanoGSM-picocell and openBSC, with
a call forwarding options and as a placeholder for other supplementary services. This enables cost saving and provides GSM features without the use of expensive telecom hardwares.

6.5 Conclusions

The work done on the thesis can be classified into the following areas.

- Surveying the 3GPP specifications for call control related to GSM Call Forwarding, CIC handling, and bearer differences in ISUP and MAP(TCAP)
- Enhancing the GSM Call Service to handle SIGTRAN family of protocols, and provide voice mail call forwarding service
- Integrating the GSM VMS TDM Call Control with GSM VMS Media Control, to provide support for voice traffic
- Implementing the working system in Erlang/OTP, and integrating it to work with existing GSM components in Mobile Arts
- Testing the retrieval and deposit of voice mail messages.

The research work adds to the existing body of knowledge of what M3UA provides. The results of the study indicate that the goal of the thesis of investigating the possibilities of creating the call control parts of a VM Call Forwarding Service in an existing GSM Call Service environment was achieved.
Chapter 7

Future Work
The integration of *GSM VMS TDM Call Control* in *GSM Call Service*, provides opportunities for enhancements and future work. The future work in in two categories namely GSM signalling, and integrating more GSM service/components.

The future work associated with signalling are CF handling, CF deflection, CCBS, and Camel, and the ones associated with integration are supplementary services like CCBS, Voice Group Call Service, etc.

- **CF Handling**
  Call Forwarding handling of mobile subscriber, includes registration, erasure, withdrawal, provisioning, activation and deactivation of supplementary services for a basic group.

  MSC can be extended, with the CF handling procedures, it also affects the modules, VLR and the HLR.

- **Call Deflection**
  Call deflection is covered in [9], it is a subset of the supplementary service provided by GSM. It is almost similar to call forwarding, but differs a bit in the call controls aspects.

- **VLR Integration**
  An actual VLR in the SS7 network can be integrated with the MSC, there are handlers in the MSC to handle the SS7 MAP traffic from the VLR.

- **CCBS**
  Call completion of busy subscriber as a result of CCBS blocking can be implemented as a CF use case. This allows the caller to be alerted when the called party has become available, and the previous call had failed to the called party.

- **Transcoder Codecs**
  This is the media control part of *GSM call service*, but also affects the call control, when different codecs are used, the transcoder can be extended to handle multiple codecs.

- **Voice Group Call Service**
  This is one of the advanced speech call items, where multiple subscribers can talk to each other, this probably wouldn’t be a near future work, but can be considered for extending the current framework.
The MG can be extended to support topology descriptor, where multiple terminations can form a topology this allows the feature for a voice call, where streams of audio from multiple termination can be part of a topology.

- Camel Phase1/Phase2 Support

Customized Applications for Mobile network Enhanced Logic was introduced for easing roaming and billing purposes, it also replaced Intelligent Network Application Part, providing greater mobility management, Camel is divided in phases, Customized Applications for Mobile network Enhanced Logic (CAMEL) phase 2 supports, call forwarding, call redirection, the MSC can extended to support CAMEL.
Bibliography


