Different mechanisms for feedback based control of operating modes and TFO/TrFO

Thesis submitted in partial fulfillment of the requirements for the degree of
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Declaration

„I confirm that this thesis is entirely my own work. All sources and quotations have been fully acknowledged in the appropriate places with adequate footnotes and citations. Quotations have been properly acknowledged and marked with appropriate punctuation. The works consulted are listed in the bibliography. This paper has not been submitted to another examination panel in the same or a similar form, and has not been published. I declare that the present paper is identical to the version uploaded."

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Outline

This master thesis is mainly focused on two new communication network operating modes named as Tandem Free Operations (TFO) and Transcoder Free Operations (TrFO).

The work has been started from the very beginning with the theoretical part explaining the background information so that in further sections operation modes can be better understood.

In **Section 1** the thesis starts with an introduction part including abstract, problem definition and the scientific method that has been followed during the thesis. Main Network divisions in the timeline and evaluation perspective are held in **Section 2**. And then the main cluster of Session Control Protocols is detailed in **Section 3**. The importance of this section has brought the necessity to be analyzed deeply so that the mechanisms of TFO and TrFO can be understood better. Afterwards the codecs and their mechanisms are explained in **Section 4**. The main focus areas of TFO and TrFO in **Section 5** have been studied including interaction and interworking scenarios in terms of operating mode requirements, codec transitions, architectural and operational aspects and structures. **Section 6** includes the reason why this work has been done. The summary and situation analysis and remarks can be found in this section. This is indeed why today's practice need as the TFO and TrFO operating modes still are not really obvious to the experts in the behavioral manner. It is hoped that this work will be enlightening some part of the road of the advancements in the communication networks.

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1 Introduction

1.1 Abstract

As the technology in mobile network advances, the mechanism search of easier, cheaper, securer and/or with better quality voice communications exponentially increases.

In the sector there are two new techniques called Tandem Free Operations (TFO) and Transcoder Free Operations (TrFO) to be said either utterly bypassing the transcoding functionalities in mobile networks or at least avoids sequential tandem transcoding activities resulted in better voice quality, less latency, less equipments to integrate and comprehensive network management.

This work mainly focuses on those two new techniques if it is possible to have the benefits and how can those be achieved.

It will indeed be seen at the end that is possible to make some extra arrangements and condition updates to the networks by integrating these two techniques for an enhanced voice and multimedia communication.

1.2 Problem Definition

The revolutionary breaking point was the release of the codecs that can be used end-to-end basis in between the fixed and the mobile networks. Accordingly the possibility of transmitting the same signal throughout the whole network without any transcoding process – fascinated the engineers.

It has been proved that such communication methods indeed can be integrated into the latest technologies. But as there are many side entities are involved it is really complicated. At start I had the following questions:

Q: Under which circumstances totally bypassing the transcoders are possible?
Q: If so, can transcoders be omitted from the mobile networks for good?
Q: Would there be the case that we still have to use transcoders?

Q: If transcoder usage is a must then are there any sequential repeating processes that we can get rid of?

The questions will be answered in the conclusion section.

1.3 Task and Requirements

The purpose of this thesis is to clarify TrFO and TFO techniques and the necessary background information with an upmost clear explanation depth level. It has also been aimed to enlighten the road of TrFO and TFO for the researchers that would like to work with and the companies which would like to integrate the techniques into their infrastructures.

In terms of understanding the background concepts and therefore the main area of interest, target audience is strongly encouraged to have a Mobile and Fixed Network, Session Initiation Protocol (SIP) and accurate IP Multimedia Subsystem knowledge. Although some of the main stream background information is studied in sections 2, 3 and 4 nevertheless it was not possible to get through all the necessary details.

1.4 Scientific Method

The scientific method that has been used in the preparation of the thesis started asking the right questions to the one of the main phenomenon of today's networks.

The perspective of TFO and TrFO research began firstly from the main source area of 3GPP technical specifications. The focus area has been studied firmly despite of hundreds of cross-references without losing the direction. In this stage the main goal, pre-requisites and pre-conditions are noted and more questions are followed to be answered later on. After having an intensive idea about the native mechanisms, the library and the resources of IEEE helped the cause by examining the newest technologies, usage areas and statistical outcomes. Approximately 75 articles (the main articles are listed in References Section) and scientific essays have been studied, compared and noted in importance order. All for gaining the enough knowledge amount in order to write a master thesis upon.

For the practical parts, the technical description documents of Telecommunication Infrastructure Vendors (listed in References Section) have been carefully detected. This has indeed the upmost importance because the thesis should also include practical aspects to be benefitted. The biggest advantage of such documentation is to bring over some points that seem unimportant in the technical specifications but indeed very essential in practical applications. Especially for the Mobile network applications of the TFO and TrFO techniques, those documentations were very helpful.

Although some technical points might seem a little bit confusing for the people that has some distance to the configuration activities, it is assured that all the aspects have been studied are used heavily in the kitchen of mobile networks integration. In other saying; this master thesis is a pure operational source and contains no junk information.

2 Evolution of Networks

The communication technologies have achieved enormous advancements through the past years. This long journey started with the invention of fax service in 1843 and has come to the point of LTE services today. Alexander Graham Bell's curiosity of hearing people from the distance and passion for that invention has brought us the vision where no man can ever imagine when the telephone was first introduced.

About 30 years after the invention, telephone become desired in every household, company and workplaces as people have seen that speaking to somebody without the need to go somewhere is indeed awesome and ease the life. The situation has caused great efforts as the demand for the services obligate the infrastructure investments, skilled people and research&development activities. Public Switched Telephone Network (PSTN) was introduced in 1878 with the logic of distant networks interconnected to each other. They needed switching services to communicate with the desired network. Switching here means making cross connections between the lines. And the switching functions were given by operator services to form the point-to-point connections and maintain the communication. In the late times of PSTN networks, the technology has come to the point of enabling voice, video and other network services with the current infrastructure. It is where the Integrated Services Digital Network (ISDN) introduced and was standardized in 1968. Different transmission mechanisms and the release of the digital platform have given an outrageous velocity to the technology research and development.

There is another person who also had begun a great achievement for first steps of telephony: Guglielmo Marconi. He demonstrated that it is possible to use electromagnetic transmission over air to send signals. His base foundings were developed parallel to the telephony networks and the scientists have done trials to send the speech signals wirelessly over the Atlantic Ocean. Later on these experiments would result in wireless networks therefore mobile phones. In 1927 the first wireless telephony service between London and New York was established and the standardization period begun.

This section includes information about PSTN, ISDN and PLMN networks, their architectural and operational aspects. It should be noted that information depth will only cover the scope of this thesis.

2.1 Public Switched Telephone Networks (PSTN)

Public Switched Telephone Networks (PSTNs) are coming from the very first version of Landline telephony systems also known as Plain Old Telephony Service (POTS). They are aggregation of Landline telephony systems that exist in different countries and the interconnections in between. Public switched telephone networks are simply created to facilitate bidirectional point-to-point circuit-switched voice communication. As PSTN was introduced with analogue communication; the method to establish a connection between two points was to tie up both ends physically creating a loop on switching boards operated by switchboard operators. As time passed PSTN became popular and started to be widely used, engineers have faced a number of problems maintaining this huge network. Immense effects of Analogue signal degradation level over the long distances, repeaters' noise creation problem and dedication of each line to only one subscriber resulted in insufficient capacity and infrastructure complications. As a solution of the problems digital communication, analog-to-digital conversion, architectural achievements such as centralized switching equipments and multiplexing methods have been introduced in order to use the communication lines in an effective way.

It is important to mention that there are significant differences between the first version of PSTN and the version that is still used today. According to the scope of the thesis – modern PSTN structure will be detailed.

2.1.1 Components and Architecture

This section includes the components and the architectural aspects of today's PSTN Networks. Illustration of the architecture and components can be seen in Figure 2.1

General Characteristics of PSTN networks are following:

- Analog signal access, 300-3.400 Hz
- Circuit-switched bidirectional connection
- Switched bandwidth of 64 Kbps

Terminals

Terminals are identified starting or the end point in the communication. Today's PSTN uses land phones (analog), Cordless Telephones (Analog with embedded A/D converter), Faxes (Digital with analog modem) and Computers (Digital) as terminal equipments in the communication.

The Access Network

Tern Access Network in PSTN refers to the local network consists of customer premises equipment (CPE) and the subscriber line.

The Central Office

The central office (CO) is the premise where local exchange switches are. The service scope of COs depends on their traffic density and subscriber number. With the new technology they serve better utilization.

Switch-to-Switch Trunks

The link between the switches is called Trunk. In case of an incoming call to a switch's subscriber, switches may establish the per call basis communication path forming one or more trunks. Trunks use Time Division Multiplexing method in order to use the channel capacity effectively.

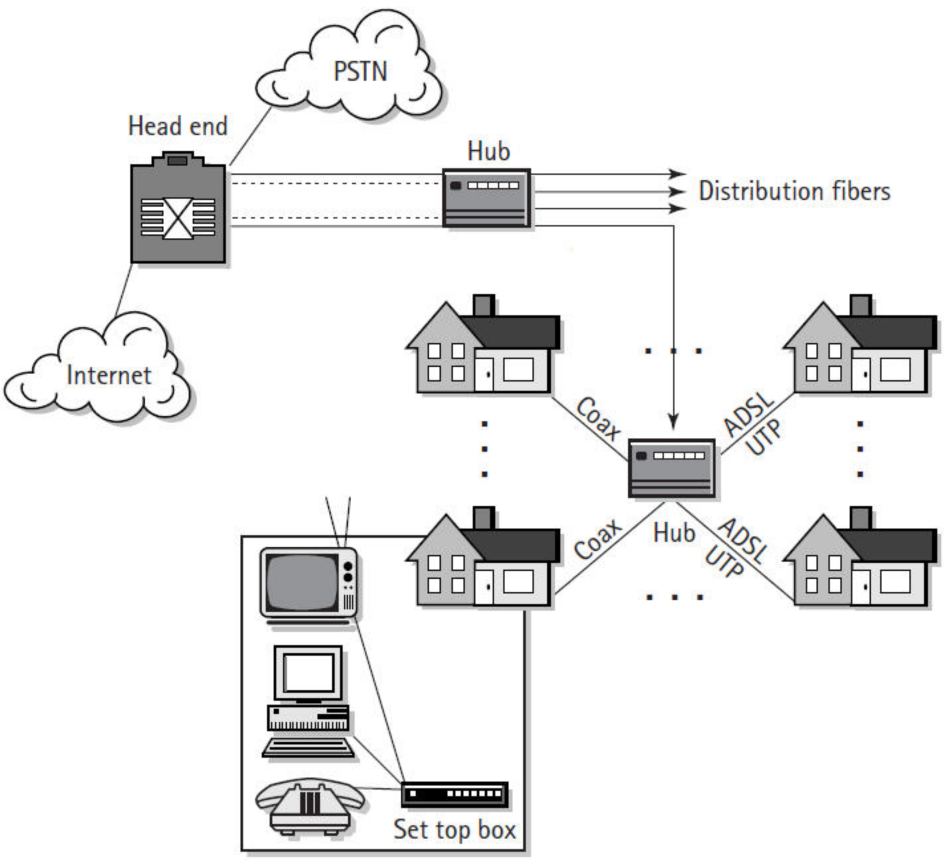

Figure 2.1 : Typical PSTN Architecture [23]

2.1.2 Signaling

As it is stated, PSTN architecture is based on creation physical/ logical loops by allowing or denying the electricity flow through the telephones. Either the traditional telephones with the hook or the modern cordless versions are designed in the ground of this fact. In the telephone networks the electricity shows itself in tone signalings such as:

Dial Tone

When the loop has been established and electricity through the phone, the most basic tone also known as Dial Tone has been released so that it can be understood that the communication subscriber line is actually online and ready for the service.

Address Signaling

Address signaling is a form of coding of the address to reach the other end terminal. Address signaling has two major classifications: Dual tone multi frequency (DTMF) and Pulse.

Dual tone multi frequency is used in DTMF enabled phones appointing different frequencies each matching a number on the key pad. Traditional Pulse telephone is generating the signals by when the movement of the wheel triggers the regulator brake and this vibration creates a frequency matching each number.

Busy

Indication signals with rather more sequential than the other tones, informing the addressed subscriber line is already online/busy.

Ringing

Ringback indication signal is used to inform the caller that physical/logical subscriber line matching is done and waiting for the response of the callee.

2.1.3 Multiplexing and Multiplexing Hierarchies

Although there are quite a few multiplexing techniques, in the practical applications single telephony channels generally use pulse code modulation (PCM) and time division multiplexing (TDM) to group 24 or 30 channels. Time Division Multiplexing (TDM) introduces digital transmission with much better bandwidth utilization.

Pulse Code Modulation

The first multiplexing first application was developed in the early 1960s for increasing the capacity of existing copper pairs between switching nodes and eliminate the noise problem. The process is based on 8-bit coding of samples at 8 kHz equals to the 64 kbps fundamental channel rate. Because the signal is quantized before the transmission medium noise signal interference is downgraded to the minimum level.

Pulse Code Modulation is used in analogue-to-digital transformation so that the benefits of digital communication can be used such as resistance to the degradation in long distances and high multiplexibility of the subscriber lines.

First the analogue signal is sampled at 8 kHz in order the sideband frequencies can be taken up to a multiple integer of the sampling frequency and with the aid of the transistor technology the amplitude points are composed for each sample. This level is called sampling. Then the amplitude levels are detected and coded into the interval of 256 levels represented by binary bits. This is Coding level. Afterwards each pulse has its own binary amplitude value and the signal is successfully transcoded to a binary stream at 64 kbps. In other saying: from analogue to 1s and 0s. The process and the intermediate levels are illustrated in Figure 2.2

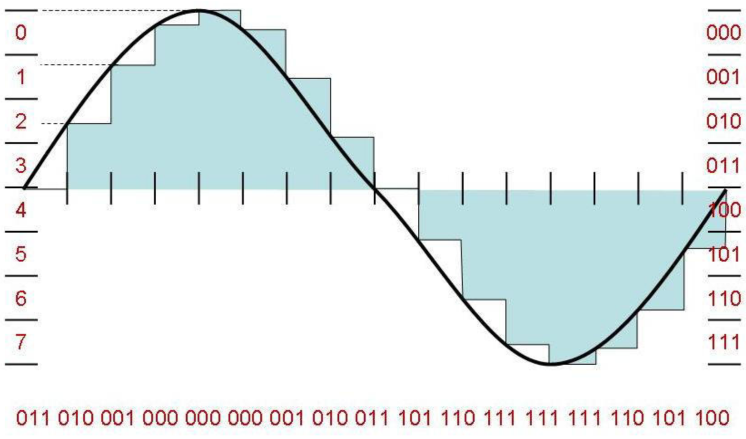

Figure 2.2 : Pulse Code Modulation [25]

Time Division Multiplexing

TDM is a multiplication mechanism that allows multiple individual signals or frames to be transmitted simultaneously from the time divided sub- channels of the communication line. Each voice sub-channel of 64 kbps is named a Digital signal Level 0 (DS0).

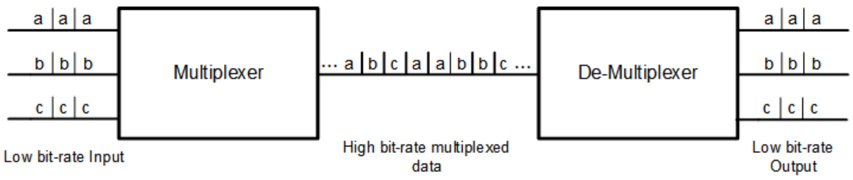

Figure 2.3 : Time Division Multiplexing

At the transmitting end, the multiplexer reads and buffers the signals which are assigned with the relevant port and/or channel number that are associated with the devices. When the buffering of the frame is done, multiplexer prepends those frames with a framing bit preseeing that the intermediate equipment may need for synchronization and buffer may be applied in the demultiplexer side.

On the receiving side the whole process gets reversed in order to get the low bit-rate signals back to their individual addresses. The illustration of multiplexing and demultiplexing events can be seen in Figure 2.3

There are two main Hierarchy Models are used in the telecommunication systems today. Synchronous Digital Hierarchy (SDH): Is mainly used in Europe and the basis signal E1 consists of 30 DS0s. Synchronous Digital Network (SONET): Is used in North America and Japan. The basis signal T1 is consists of 24 DS0s. The hierarchy can be seen in Figure 2.4.

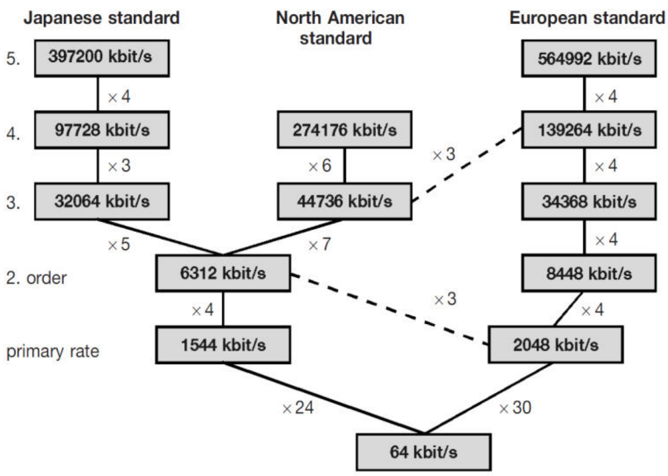

Figure 2.4 : Multiplexing Hierarchy SDH and SONET interaction

2.2 Integrated Services Digital Networks (ISDN)

The technological achievements have brought us to the point to be eligible of the transmission of not only telephony signal but computer data signals, including faxes, files, web pages, sound and pictures. And all those entities must be represented by a common integration technology such as packet-switching. On the other hand it was clear that although it works somehow, transmitting those data traffic with the desired velocity and reliability was not possible with the analogue medium conditions of PSTN.

It is very expensive and complicated to renew and support the analogue infrastructure with optical fiber transmission cables. SO there are new hardware such as digital intermediate equipments and rapid switching centers for a faster, more reliable and integrated communication could have finally been used in the new networks. Integrated Services Digital Network (ISDN) by CCITT is released in 1988. ISDN is an assembly of communication standards; allowing packet-switched voice, image and data communication via digital medium and PSTN infrastructure.

As the focus of the master thesis Broadband-ISDN networks will be outlined in this chapter with preventing the unnecessary details not to create confusion. In terms of the protocols that are used in ISDN only ISDN user part (ISUP) will be detailed.

2.2.1 Components and Architecture

The Integrated Digital Network generally consists of local switching terminals equipped with digital transmission links carrying 64 kbps digital channels multiplexed on to copper pairs, coaxial pairs or optical fiber bearers.

Terminal Adapter (TA)

Terminal Adapter is a converter device that converts standardized ISDN signals into the entities to non-ISDN devices to set up the interconnection between the networks.

Terminal Equipment Type 1 (TE1)

TE1 maintains the interface between the ISDN network and CPEs such as telephones, personal computers, and fax machines.

Terminal Equipment Type 2 (TE2)

TE2 maintains the connectivity with the CPEs that cannot directly face the ISDN network such as analogue phone or modem with the ISDN network.

Network termination type 1 & 2 (NT1 and NT2)

A compact physical connection that is used connecting the customer site to the local loop providing two pairs of connection to the customer site and a pair connection to the network

2.2.2 ISDN Channels and Implementations

The ISDN uses dedicated channels to the specified purposes. There are mainly two channel types that are used: Bearer type 64 kbps channels (B-Type) are used for only the data and voice traffic purposes carrying user service information. Delta type 16 or 64 kbps channels (D-Type) are mainly used in signaling and control purposes but can also be used for data traffic transmission.

ISDN implementations are mainly defined according to the customer needs but generally formed of those known channel types;

Basic Rate Interface (BRI)

BRI interface is formed of two Bearer channels of 64 kbps and one Delta channel of 16 kbps.

Primary Rate Interface (PRI)

Primary Rate Interfaces are mainly focused on Telecommunication suppliers and consists of numerous channels and the speed is depending on the Hierarchical order that is used. For Europe PRI Interfaces consists of thirty Bearer and one Delta Channels equals 2.048 Mbps and in North America and Japan PRI interface contains twenty three Bearer and one Delta Channels equals 1.544 Mbps.

Broadband Integrated Services Digital Network (BISDN)

BISDN is used as an interface within the network backbone structures working over ATM access technology.

2.2.3 ISDN User Part (ISUP)

The integrated services digital network user part (ISUP) is indeed a part of Signaling System No. 7 and will be well detailed in the next chapters. ISUP simply the signaling protocol which supports the establishment, maintenance and tear down of voice and non-voice calls over circuit-switched connections between ISDN network points of digital subscriber access lines.

Basic ISUP Call Control

When a call is triggered, the originating Service Switching Point (SSP) transmits an ISUP initial address message (IAM) to book if there is an idle trunk circuit to the destination switch. The IAM message includes the originating point code, destination point code, circuit identification code, dialed digits and sometimes the calling party number and name.

After the destination switch detected the dialed number, it is inspected if the line is available for ringing. The destination switch rings the called party line and transmits an ISUP address complete message (ACM) to the originating switch to indicate that the remote end of the trunk circuit has been booked. The process may continue till all the trunks on the way are fully reserved. The Basic call setup and release with Initial Address Message (IAM), Address complete message (ACM), Answer message (ANM), Address complete message (ADM),Release (RLS), Released (RLD) and Release Complete (RLC) messages can be seen in Figure 2.5

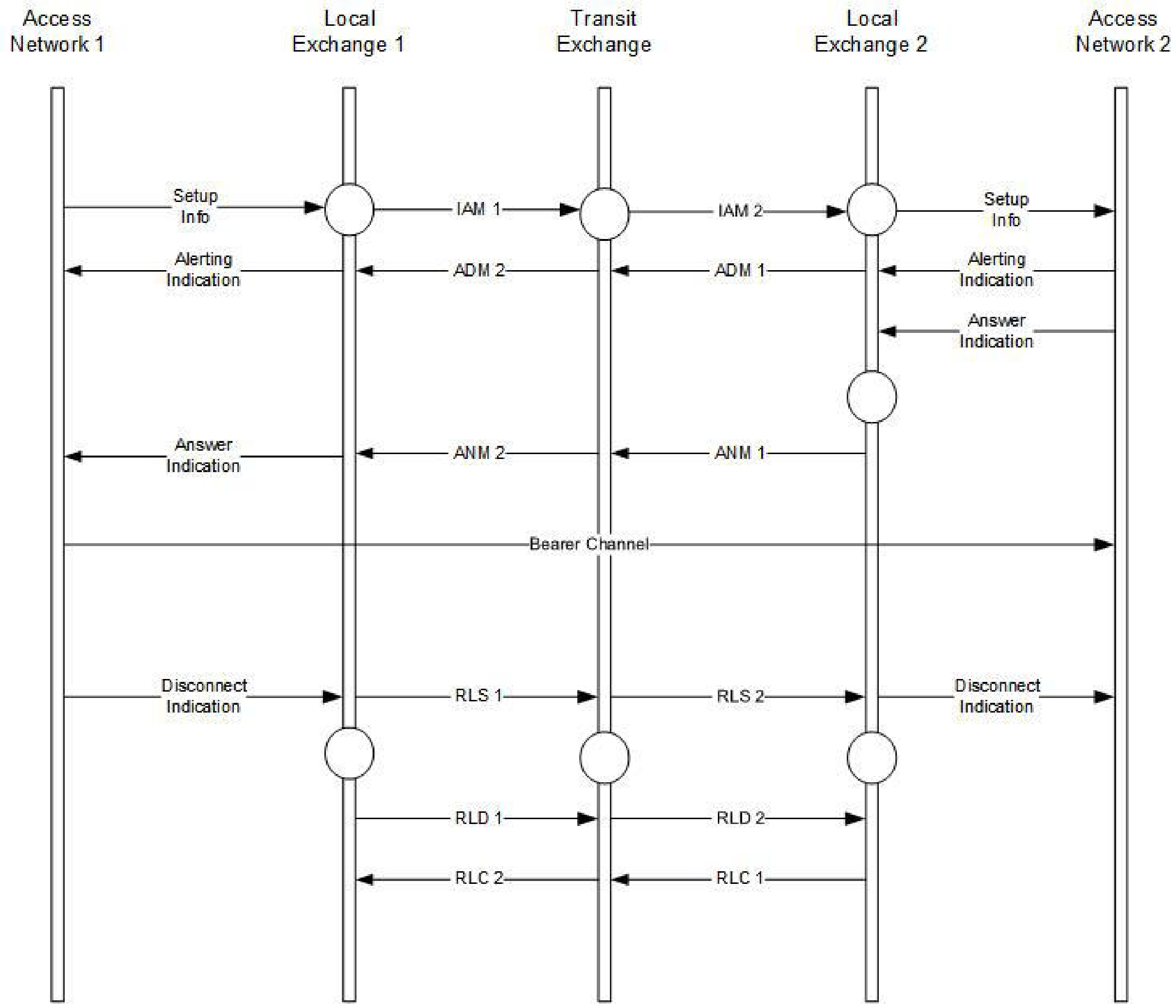

Figure 2.5 : Basic ISDN Call setup and Release

2.3 Public Land Mobile Networks (PLMN)

The public land mobile networks (PLMN) correspond to any wireless communication system aimed for use by the mobile subscribers. The system is not only a standalone service but is connected via several transmission technologies and interconnection facilities to PSTN and ISDN networks. The most popular services of PLMN are voice and short message service (SMS) communications. But nowadays as the new technologies Wireless networks still achieved, internet usage via mobile stations such as cell phones and tablets etc. are increasing immensely.

The idea of PLMN networks is that supplying approximately same level of service as in a fixed network but via a different access method. Sometimes it can be seen as a challenge in regions where the terrain is irregular, where base station sites are hard to find and maintain, and in urban environments where there are numerous obstructions such as buildings, and myriad sources of radio-frequency (RF) radiation that can cause noise and interference. Most systems today use digital technology rather than the older analog methods [25]. As the scope of the work GSM and UMTS networks will be illustrated and outlined.

To have a better understanding in the next sections, the following terms entitled to Mobile Networks, need to be explained:

Noisy scenarios: A mobile communications scenario is inherently noisy, with high and very variable noise levels (many different situations: public places, car cockpit, etc., hands-free operation mode, etc.

Speech codec (encoder-decoder) distortion: Standard codecs are designed to work at specific bit rates while maintaining the perceptual quality as high as possible. The distortion introduced by the codec, which becomes higher when the bit rate is low.

Transmission errors: Due to the unreliable nature of the radio frequency channel, transmission errors are much more influential than fix wire-links.

The noisy speech problem has been addressed in different ways in the frame of Mobile Network environment: speech enhancement, robust parameterizations, model compensation etc. But the attention should be on more specific of the wireless transmission problems: speech coding distortion and transmission errors [31]

The Evolution of PLMN Networks in time as follows:

- **GSM Phase 1:** The dominance of GSM Phase 1 Network is that ISDN Rapid Switching entities here are enhanced to Mobile Switching Centers (MSC). Network illustration can be seen on Figure 2.6

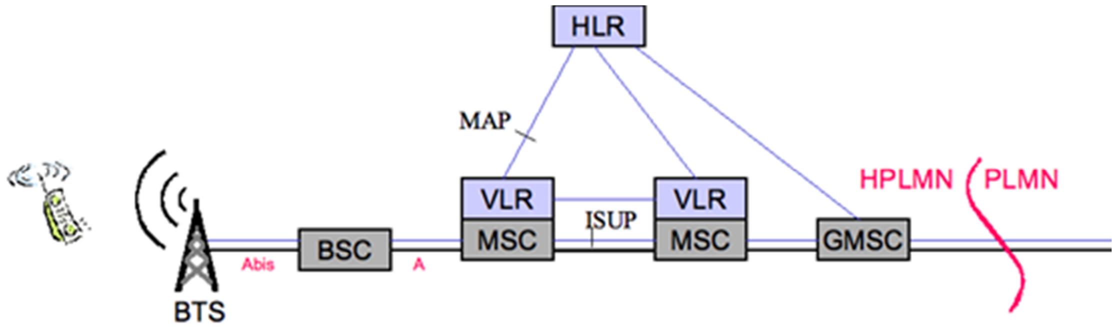

Figure 2.6: GSM Phase 1 Network Topology [28]

- **General Packet Radio Service (GPRS):** The revolutionary efficiency discipline of GPRS Networks has allowed the dynamic allocation of offline transmission capacity to be temporarily addressed to the current subscribers. So that the capacity can be used in a very effective way. On the other hand internet has been introduced to the mobile networks adding the ability of dealing with data packets. Serving GPRS Support Node (SGSN) and Gateway GPRS Support Node (GGSN) is released.

- **Universal Mobile Telecommunication System (UMTS) Rel. 99:** The significance of this release is that the previous architecture stays fixed and new radio domain Node B and Radio Network Controller (RNC) are introduced.

- **UMTS Rel. 4:** UMTS release 4 is introduced as using the Bearer Independent Core Network (BICN) separating user plane and control plane from each other to allow several protocols to be used within the network.

- **UMTS Rel. 5:** The biggest achievement of UMTS Release 5 is that IMS has been introduced to the mobile networks. In this release Mobile Switching Stations (MSS) and Visitor Location Register has been introduced separately from Mobile Switching centers (MSC). Network illustration can be seen on Figure 2.7

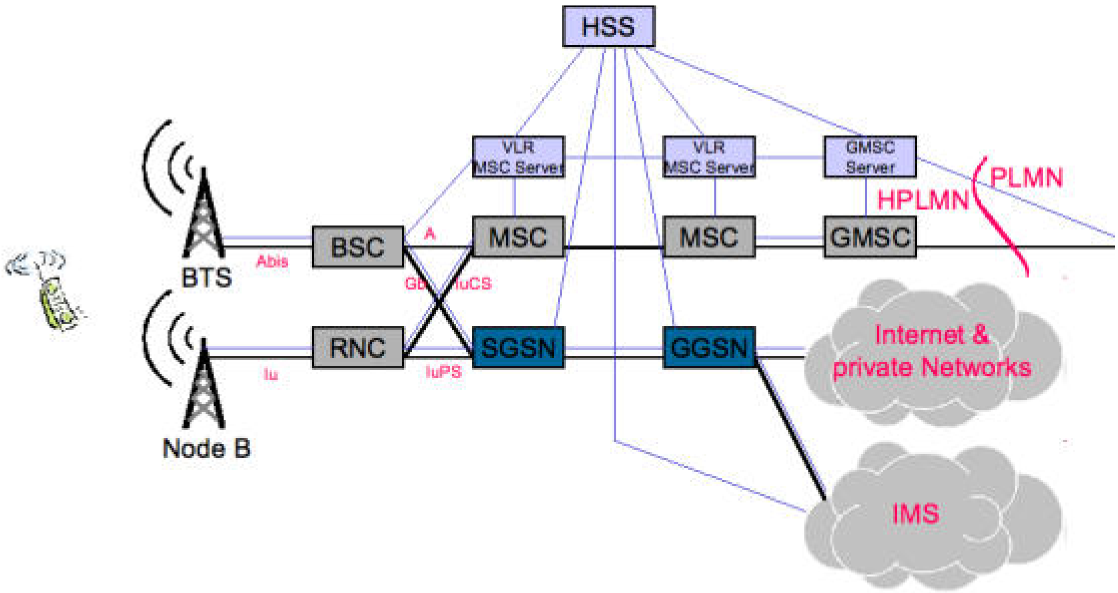

Figure 2.7: UMTS Rel. 5 Network Topology [28]

3 Session Control Protocols in Core Networks

3.1 Introduction to Session Control

A Session in communication networks refers to each interaction of information interchange in the network that has stated beginning and end. Sessions are initiated by signaling message flows and an established session may involve more than one message in each direction. Accordingly there are three sets of bidirectional message flows that every session includes: As first Session Signaling Messages to create, modify and terminate the processes. Secondly Data Streams are the actual data that are subject to communication and Media Control Messages are used to hold, create and send reports to the transmission feedback mechanisms so that the session can take place. Whole means of communication is nothing more than a string of sequential sessions.

In today's modern networks, it can be observed that the data stream types may vary depending on the purpose and abilities of the network such as application sharing, internet telephone calls, multimedia distribution, multimedia conference, instant messaging and internet surfing. All Networks tend to make sure that the communication from one end to the other is fully successful. This requires task distribution, functioning in an order and a perfect harmony between subsidiaries. In order to accomplish the overall controlling of the sessions, various network elements, endpoints and mechanisms are using one or more session control protocols. In terms of hierarchical OSI layer model session control protocols run on top of various different transport and network protocols on layer 7.

Detailed information about session control protocols can be found in following sections of the chapter.

3.2 Signaling System No.7 (SS7)

Signaling System No. 7 (SS7) is standard for telecommunications defined by the ITU-T. The standard is an assembly of the procedures and protocols of network elements in the PSTN networks aiding to exchange information over the digital medium to work with wireless and wireline call setup, routing and control.

The SS7 common Channel signaling system protocols are used for:

- Basic call establish, management and tear down
- Wireless services such as wireless roaming and mobile subscriber authentication
- Local number portability (LNP)
- Call forwarding, calling party name/number display and three-way calling
- Efficient communication

3.2.1 Components

SS7 system consists of mainly 3 entities that are uniquely numbered throughout the network:

SSP (Service Switching Point)

Actual switches in the task of establishment and tear down of the calls communicating with the other SSPs in order to achieve this task. Like also the other components they have their own authentication and decision in routing the packets through the network.

STP (Signal Transfer Point)

Signal Transfer points a whole packet switching mechanism forwarding and routing the messages according to the routing information that exist in SS7 message's relevant header

SCP (Service Control Point)

Service control points are often co-located with the STPs providing error correction and retransmission capabilities and addressing the link/communication failures in the network.

3.2.2 Protocol Stack

It is important to have an overview on the protocol stack of SS7 comparing the OSI layers. SS7 Protocol stack can be seen in Figure 3.1

Some important outcomes are following:

Message Transfer Part Level 1: Is on the same task area as OSI Physical layer, defining the physical medium transmission characteristics and interfaces.

Message Transfer Part Level 2: Corresponding Data Link layer of OSI ensures reliable end to end transmission, error detection and error correction of data streams through the signaling path.

Message Transfer Part Level 3: Acts like the Network Layer in OSI aiding end to end routing and re-routing the packages across the network.

ISUP : Referring to the Section 2.2.3 ; ISUP simply the signaling protocol which supports the establishment, maintenance and tear down of voice and non-voice calls over circuit-switched connections between ISDN end points of digital subscriber access lines.

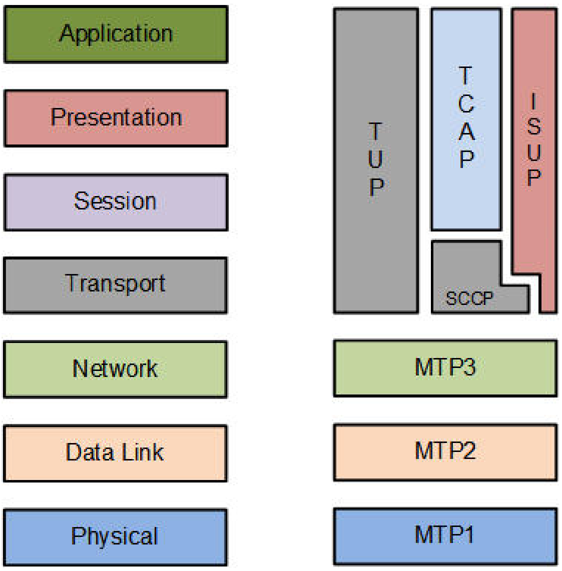

Figure 3.1: SS7 protocol Stack [27]

3.3 H.323

It was around the time when voice over Internet protocol (VoIP) was becoming a common alternative to traditional Plain Old Telephone System (POTS) phones since the new era of deployed IP Networks offered a low-cost but low-quality voice communication over direct access to the web.

Voice over Internet Protocol end-user applications are still based on the actual infrastructure by utilizing POTS terminals plugged into analog-to-digital routers and telephony software installed on a PC. Unlike the POTS networks that have almost no initiative for the end users, those telephony applications allow user-level control of the call at the same time having the functions of presenting and receiving the multimedia streams. The biggest advantage of VoIP discipline besides lower quality of voice in the communication is that it keeps the application functionality reasonably separate from the lower levels, application hardware or software is easily upgradeable and user privileges changeable, with significantly minimized upgrade costs. Recalling very well-known situation when ISDN was introduced, the costs of hardware and software upgrade were reasonably high.

H.323 is an umbrella protocol standard by ITU-T that includes a set of protocols and services setting ground for voice over Internet Protocol communication call establishment, management and termination. It specifies the procedures, protocols and components that provide multimedia communication services: real-time audio, video, and data communications over packet networks. H.323 is part of a family of recommendations called H.32x that provides multimedia communication services over a variety of networks. A big advantage of H.323 was being one of the first available disciplines, defining the basic call model as well as futuristic featured-services, needed to address business communication expectations. These open-minded predictions have caused the situation that even nowadays H.323 is widely deployed worldwide by service providers and enterprises for both voice and video services over IP networks.

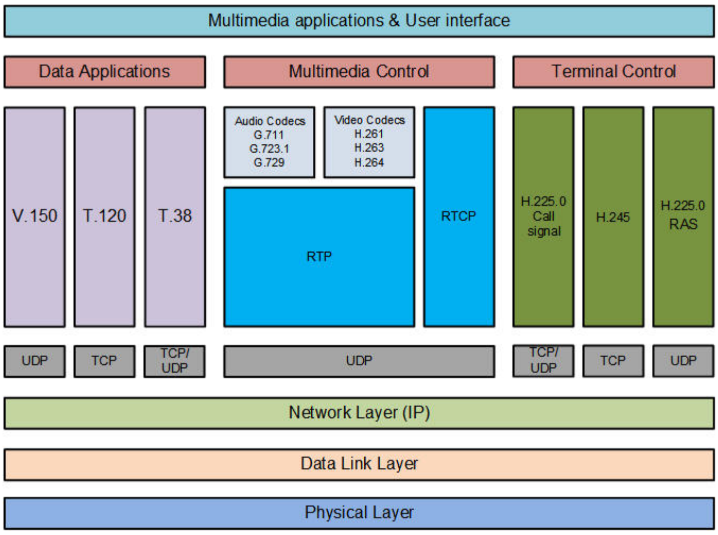

Figure 3.2: H.323 OSI Layer mapping and associate protocols

The protocol stack structure of H.323 can be also seen in Figure 3.2. It should be defined that the several other protocols within H.323 also are usable through mappings to OSI's presentation and session layers. The H.323 presentation-layer services include codecs that are used to encode and decode the voice communications. Each codec runs a different coding rate, specified to different levels of voice quality. On the other hand video codecs are also available, including H.261, H.263 and H.264. An important feature of the protocol is that keeping the codecs at the presentation layer allows VoIP systems to change data coding rates to match network congestion or update newly released codecs without modifying the user application.

The standard of H.323 specifies five different components in its architecture to form a complete session control:

- **Terminals:** Endpoint/ User end equipment in order to manage multimedia streams as applications. Terminals can either be applications of soft phone or mobile smart phone and/or PDAs that used for bidirectional communication. The several other protocols that H.323 includes are used in order to maintain compliance to different kinds of terminals.
- **Gateways:** Consists of Media Gateways (MGW) and/or Media Gateway Controllers (MGC) that in some cases may coexist. Forms bridge connections between different kinds of networks. Connection process

includes data delivery, non-media related functions and protocol translations for call establishment and management.

- **Gatekeepers:** Optional components to take role on addressing terminals and gateways, admission control, bandwidth management and charging. Gatekeepers provide direct end-to-end call services as well as maintaining call-routing services.

- **Multipoint Control Units (MCU):** Engagement of multi-party call sessions such as conference calls with two or more participants. MCUs have the task of negotiation between endpoints to determine the coding for audio and video in communication.

- **Peer Elements:** Equipments that are on duty of addressing information interchange and maintaining authorization within administrative domains (AD)[1]. Peer Elements are also used in compressing the routing information in terms of making routing tables smaller and easy to transfer. If the Peer Elements are located between two different administrative domains then they are specifically called as Border Elements.

3.4 Session Initiation Protocol (SIP)

SIP is an application-layer control protocol that can establish, modify, and tear down multimedia sessions such as telephone calls, tele-conferencing, image transmission and etc.

SIP will only be outlined here as the main focus will be TFO and TrFO disciplines. For more information about SIP please refer to RFC 3261 by IETF

[1]Administrative Domains are specified small-scale networks that have a routing characterization as whole.

3.4.1 Main Functionality

As the invention idea SIP had been created in order to maintain multicast backbone project at first. Afterwards used for conferencing and VoIP applications and finally became a standard in 1999.

SIP is created with following goals:

- Flexibility in Transport Protocol – to be able to operate via reliable or unreliable protocols
- Routing ease in direct or with proxy
- Extensibility to the new applications
- Personal Mobility

In SIP functionality users are often represented by a generic address identifier. To provide mobility and reachability the address identifier consist no location information at all. SIP is able to operate five basic tasks [29]:

Locate the user: Determination of the location of session partner terminal

User availability check: Determination of the abilities and characteristics of the session partner to participate the session

User capabilities check: Determination of the media and media parameters

Session setup: Triggering the intermediate entities, ringing the session partner, establishment of session parameters at both side

Session management: Including transfer and termination of sessions, modifying session parameters, and invoking services.

Session handling of SIP is pretty justified for all kinds of sessions that are competent to be named as a session as SIP mechanism do not carry session type information within the message body. As SIP is HTTP and SMTP based, all the necessary information carried within the message body as an attachment. The most popular sessions are voice or video sessions, but also instant messaging and/or online gaming applications are also handled by SIP. If the compounds of the service such as header fields and options are correct then there is no reason why any service doesn't work.

3.4.2 Principal Operations

The operation will be given over an example is about two SIP users (Alice and Bob) willing to communicate with each other over the internet. Users essentially may have a VoIP client installed on their smart phones or a SIP client (softphone) on the PCs with the assumption that both persons have already an account (a SIP identity) on those clients in the form of a SIP URI. The SIP URIs of Alice and Bob are like in this example “sip:alice@atlanta.com” and “sip:bob@biloxi.com”

The SIP URI consists of a user- and a domain- or host-part separated by the “@”-sign indicating a belonging to a certain domain as in this example atlanta.com and biloxi.com. The illustration of the basic operation can be seen in Figure 3.3 [29]

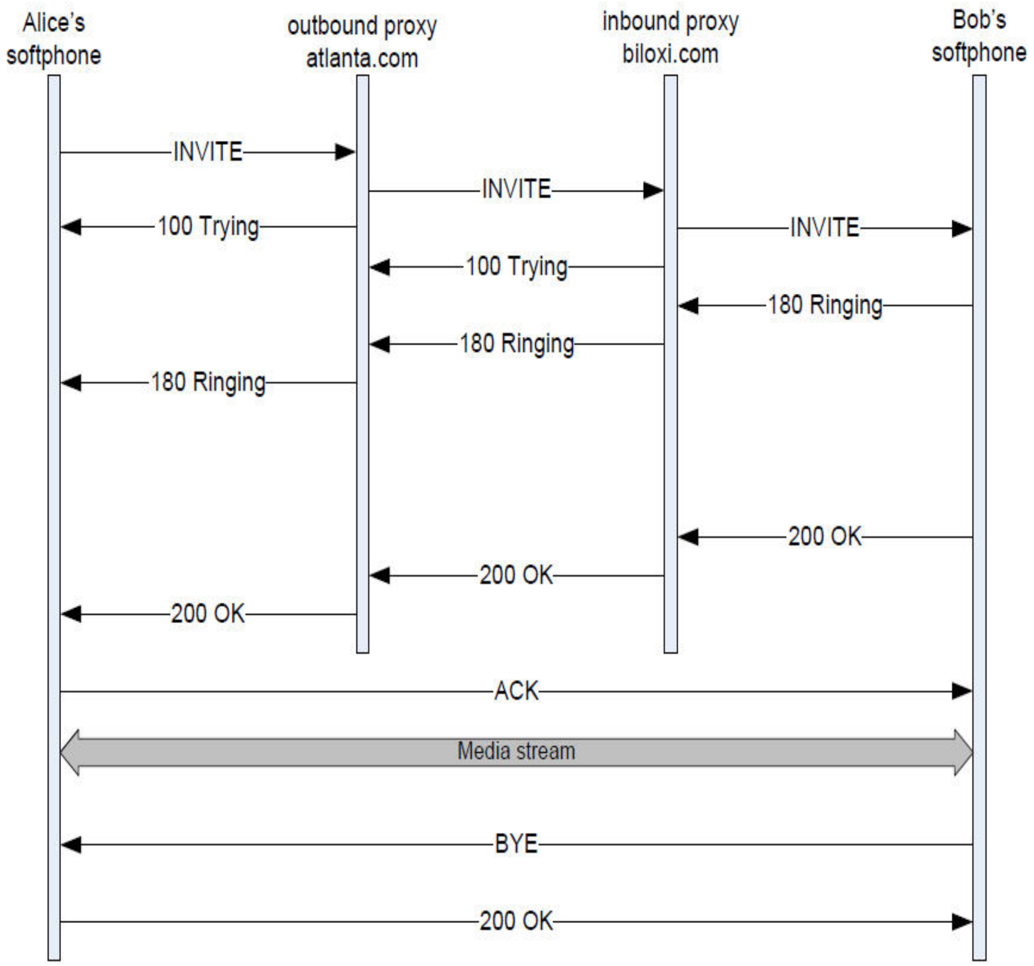

Figure 3.3: Basic SIP operation

3.4.3 Session Description Protocol (SDP)

The Session Description Protocol is a text based, layer 7 application used to define and describe the multimedia sessions. In the SIP session setup both end entities indicate their capabilities and willingnesses, supported media formats, respective transmit capabilities and receive address/port information via sdp protocol. An example to sdp protocol is shown on Figure 3.4

In the figure SIP version number, via headers, to header, from header, Call ID, Transaction ID: CSeq, Contact, Content type and content length information can be seen and handled from the text structured sdp printout.

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP server10.biloxi.com
   ;branch=z9hG4bKnashds8;received=192.0.2.3
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com
   ;branch=z9hG4bK77ef4c2312983.1;received=192.0.2.2
Via: SIP/2.0/UDP pc33.atlanta.com
   ;branch=z9hG4bK776asdhds ;received=192.0.2.1
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 INVITE
Contact: <sip:bob@192.0.2.4>
Content-Type: application/sdp
Content-Length: 131
```

Figure 3.4: Basic SDP message

3.5 Bearer Independent Call Control (BICC) Protocol

Since the beginning of the 2000s the trend of growing bandwidth demand and number of users in the networks had been obvious for all the network operators. Numerous published reports have shown that Internet and relevant ATM/IP traffic within networks override the traditional telephony traffic in bandwidth and demand. This situation had triggered the convergence between data and voice networks.

The ATM/IP traffic growth was significant for all the operators. Capacity of the current infrastructure resulted in serious inadequacy on daily turnovers of the networks. On one perspective operators had to spend major amounts to extend their actual TDM networks to accommodate the growth. However there was no logic in investing old telephony networks whereas the trends of technology demanding packet networks implementation. Accordingly operators tended to invest deploying packet networks besides using their TDM infrastructure. By doing so an there was also interworking procedure needed to connect the networks all together. As it is mentioned on the previous section the Session Initiation Protocol, based on a crucially different call model than the standard protocols, was not to handle this situation therefore the need of a new operating-level protocol was obvious. This new work had to [9]:

- be based on ISUP call model, to be fully compatible with existing services,
- be independent of the underlying technology,
- re-use existing signaling protocols to establish the communications within networks

Beginning of the year 2000 ITU-T has developed a new approach to maintain the feature-rich services to be offered supporting packet networks using the standardized PSTN and/or ISDN protocols. The main idea of getting communication independent of bearer is a cost efficient and qualified way of transmission by enabling the codec negotiation functionality independent from bearer technology, which is commonly ATM or IP, named as Bearer Independent Communication Control (BICC)

The Bearer Independent Call Control (BICC) is a network level call control signaling protocol based on the existing narrowband ISUP call model specifications [6]. BICC mechanism functions by separating call control and bearer connection control, co-operating with application transport mechanism (APM) transmitting the specific bearer signaling independently of bearer establishment process. By all means BICC aims to remove the disadvantages of SS7 protocol, where bearer and call control mechanism is combined.

The ITU-T has released two versions of BICC. Capability set 1 (CS1 BICC) including the specializations of Forward and Backward backbone network establishment, Narrow band services support, interworking with ATM Adaptation Layers (AAL1 and

AAL2) and Backbone Network Connection (BNC).As through the needs of the industry and technological advancements Capability Set 2 (CS2 BICC) has been released in year 2002 having additional specifications to the protocol such as Local exchanges support, Decomposed multifunctional ISN, call mediation node function, IP bearer support and IP related equipment and services such as IP tunneling, MSC and GMSC servers. In the following sections technical details of BICC Capability Set 2 will be given.

3.5.1 Protocol components

Bearer Independent Call control Protocol, as an application layer discipline, runs in co-operation of several other protocols and mechanisms to be defined for a fully understanding.

- **IP Bearer Control Protocol (IPBC):** IP Bearer Control Protocol is an sdp based protocol to maintain establishment of IP bearers on call basis. IPBC is used for bidirectional transmission of media stream specifications such as RTP port numbers as well as IP addresses of source and sink nodes.

- **Bearer Control Tunneling Protocol (BCT):** Key component of IPBC Protocol used for tunneling mechanism between BICC and Call Bearer Control (CBC) interfaces.

- **Signaling Transport Converter (STC):**Is an internal CSF function that converts primitives from lower and higher layers and their parameters in accordance of compatibility requirements of the other layers

- **Stream Control Transmission Protocol (SCTP):** Has grown from investigations into the limitations of TCP and research into distributed computing applications [14]. Used in reliable transmission of N-ISUP information throughout the IP network by retransmission of data – also supplying error checking, sequencing and keep alive mechanisms.

- **Application Transport Mechanism (APM):** Specifically used by the applications sending application-specific data throughout the network using either via a separate APM message or call control (CC) messages' optional parameter part. This specification enables the network to get rid of additional data and frame structures maintaining less complexity. APM is also needed to transport bearer-related information such as information elements (IE) of BNC ID and BIWF address in a BICC network.

- **Bearer Internetworking Function (BIWF):** A functional composition service of CSF, Media Control Function and Media mapping and switching function (MMSF) which is used to have simple media gateway (MGW) functioning.

3.5.2 BICC Architecture and Functionality

It can be said that BICC does two main functionalities. First, to set up a connection over the packet networks - depending on the packet network type - in order to be able to transmit ISDN/PSTN media streams (voice and/or video signals) and secondly call management over the bearer that has been set up. Basic functional BICC network architecture can be seen in Figure 3.5

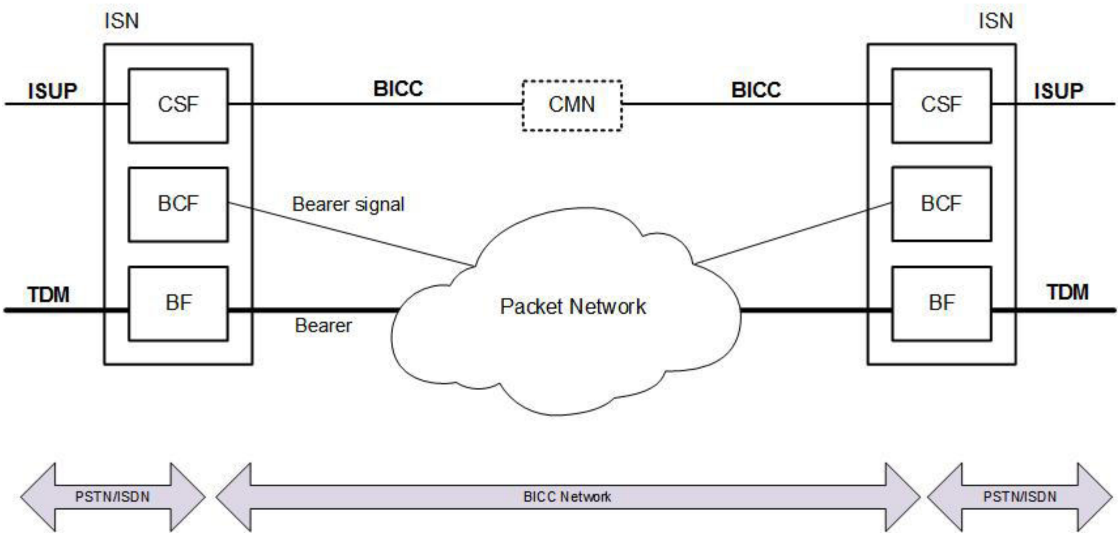

Figure 3.5 : BICC Network Architecture

In terms of BICC architecture, the technical specifications assume that BICC network starts and ends with Serving Node equipments (SN).Various networks equipments are called only when they have common Bearer Control Functions(BCG).In terms of task divisions there are three kinds of serving nodes;

- **Interface Serving Node (ISN):** Provides an interface to the non-BICC networks.
- **Gateway Serving Node (GSN):** Provides gateway function between two BICC networks interconnecting them via a single transmission line for both GSNs.

- **Transit Serving Node (TSN):** Maintains transit functionality between ISNs and GSNs. Very important aspect for two or more BICC networks communicating with each other – supplying services transparency and flexibility in terms of ISDN/ISUP networks. Supplies Intelligent Network (IN) services and packet type conversion between different packet networks.

For two or more networks that have user plane compatibility problems in case of one or more TSNs do not include any internal user plane functionality. Under these terms internetwork connection may need a Call Mediation Node (CMN) – a special type of TSN including specific user plane options specifications without associated Bearer Control Function (BCF) – to overcome the issue. On the other hand CMNs are also useful relaying the BICC protocol signaling and indeed more functional in large BICC networks.

An overall demonstration of serving nodes interacting between two BICC networks can be seen in Figure 3.6

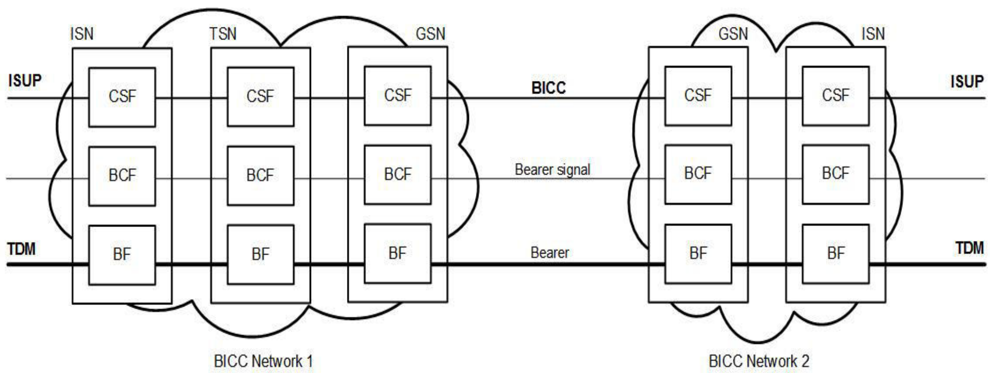

Figure 3.6 : BICC Serving Nodes

Within the capability set 2 – distinguished functionality of serving nodes had been released. According to the release there are three main parts in an SN. **Call Serving Function (CSF)** corresponds control servers and is used to manage ISUP, ISDN and

BICC call signaling per call basis – providing the transform service via STCs. **Bearer control function(BCF**) is used to establish and tear down bearer connections in packet networks and at the same time controls bearers in order to use the convenient signaling and **Bearer function (BF)** is on direct touch to the trunks, providing conversion of data streams from TDM trunks to packet networks and vice versa.

Functionality of the physical entities requires always a wide perspective in the matter. In functionality perspective a serving node has 3 divisions. Central control Unit (CCU) executes tasks of CSF and some additional side protocols to achieve call control and BICC signaling. Bearer control unit (BCU) does all over BCF functionality with some extra entities managing bearer control. Media Gateway Unit (MGU) was the first foundation function of today's modern MGWs achieving more or less the same tasks on switching, transforming and aiding to route the multimedia data streams through IP packet networks. Divided functionality of SNs can be seen on Figure 3.7

Now that it can be seen that in BICC architecture majority of the protocols works on horizontal – backwards and forwards – but a must-to-mention point is that some protocols also work in vertical plane. Call-Bearer control (CBC) functionality hence is candidate for H.248 or Megaco protocols to fulfill the task of synchronization and communication between BICC signaling and the bearer control signalings.

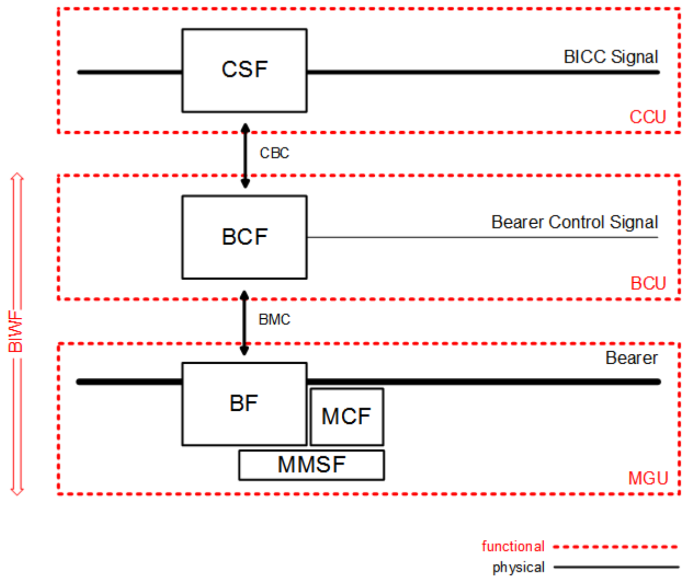

Figure 3.7 : Decomposed SN physical and functional perspectives

3.5.3 BICC Messages and Parameters

Protocols in general meaning run on controlling signals and controlling signals consists of many protocol data units (PDU) to achieve data exchange and various other functions.

Bearer independent call control protocol consists of five functional parts. A generic BICC pdu structure can be seen in Figure 3.8. **Circuit Identification Code (CIC)** is the code number in N-ISUP networks that each PCM bearer channel is one-by-one allocated. CICs are identifying the source bearer to get marked so that signaling messages can be associated to the specific calls. **Message Type Code (MTC)** related to the PCM networks, getting the same value in BICC as in the PCM networks defining the function and the format of each PDU. One fact also should be stressed here that although MTCs being almost identical in ISUP and BICC – some of the MTCs are having no correspondence in BICC network as through only supported in ISUP. **Mandatory Variable Part** contains the pointers that are used to define the start point of each parameter. **Optional Part** consists of fixed and/or variable length parameters in order to be filled by additional features.[2]

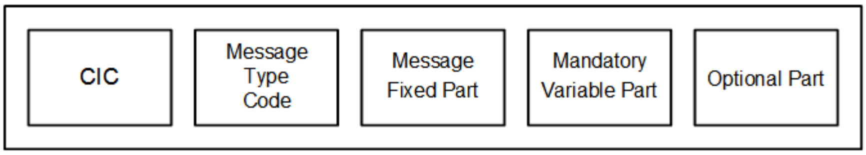

Figure 3.8 : BICC PDU structure

3.5.4 BICC Operation

In this section, a basic call setup operation in Bearer Independent Call Control Protocol communication will be explained getting from the beginning to the end in terms of a BICC network's perspective.

BICC protocol is triggered by an IAM message indicating bearer and call SETUP request coming to one of the ISNs throughout the network. The logical order is that the call setup is done before bearer setup. In terms of a BICC network between two ISDN/PSTN networks, the mission will be successfully accomplished when the receiver network is

[2]Features that have only been supported by BICC or ISUP

triggered by an incoming bearer and call SETUP signal with full set of parameters. BICC call and forward bearer setup flow can be seen in Figure 3.9

Call establishment

Call establishment is the process where two end ISNs agreed to start the protocol process. Sender ISN send the call SETUP request over the packet network to the relevant TSN in order to be addressed to the receiver. A call SETUP request is then responded by a call SETUP confirmation to define the packet network addressing information especially when BICC is used for codec negotiation. The main logic of call establishment is to set up necessary channels to negotiate over parameters for a bearer establishment.

Bearer establishment

BICC supports two traditional bearer setup methods depending to the direction of call setup. If the bearer establishment signaling path is in the same direction as call setup - this kind of bearer establishment called forward bearer setup but if the bearer establishment is on the opposite direction of call setup – this is called backward bearer setup. One big characteristic of backward bearer setup is that bearer SETUP trigger comes from the receiver end. Nevertheless backward direction is limited to support TDM network applications.

Forward direction bearer setup situation can be explained in other words that the same direction with IAM sending path where bearer control unit (BCU) ID and media gateway function characteristics are being transferred and by an APM response. Forward direction bearer setup is most suitable for codec negotiation procedure in IP networks. The signaling terminology thereafter the bearer SETUP request follows with a bearer CONNECT confirmation as this signaling is a precondition in case end user ringing without proper communication establishment[3]

Notification of Bearer Setup

Notification of bearer setup is a pre-check confirmation for some of the internal protocols – that bearer is fully setup and ready for transmission of some critical call control messages. Literally the procedure functions that way;

---

[3]This situation is called - Speech Clipping

At the point that TSN has the bearer SETUP request from originator node, it sends an ISUP continuity message COT to the receiver node in the meaning of originator side is now agreed and took actions in order to build the bearer with all parameters exchanged beforehand. The procedure becomes then fully adopted when receiver ISN sends a COT message back.

Next network trigger (Outgoing bearer and call SETUP request)

Next network trigger – is the last session of BICC operation when successful call and bearer have set up and originating ISN has its ISUP COT confirmation message. ISUP networks are then fully operative through packet networks.

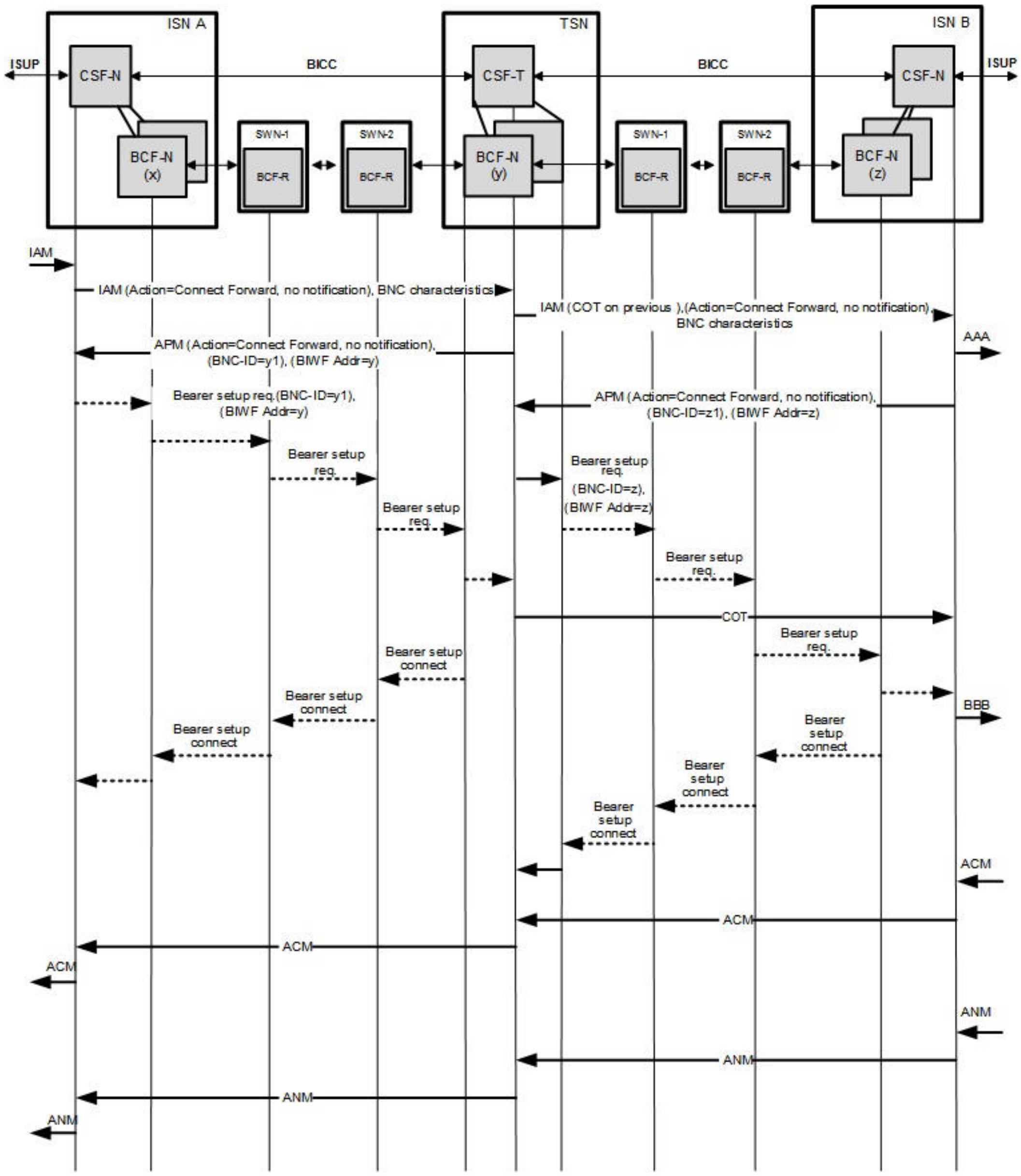

Figure 3.9 : BICC call and forward bearer SETUP flow [5]

3.5.5 Out-of-Band Transcoder Control Functionality (OoBTC)

Out of band transcoder control is an optional BICC CS2 capability – allowing control servers in mobile networks to negotiate codec and codec parameters out of band per call basis.

In mobile networks practically both end users in the network, which are supporting different codecs, can only communicate with each other after at least two times transcoding (coding and decoding) processes. Eventually when signal is encoded and then decoded several times, it gets damages and has a lower quality. Importantly noting that on theoretical basis (however transcoding facilities are supported to be not very harmful) in terms of real mobile networks, we get a lot of equipments and far more quality discrepancy.

Out-of-band signaling generally describes the signals (between two devices) are sent via a path or method different from the primary communication. OoBTC is used before bearer SETUP attempt to establish UE-UE transcoder free operation (TrFO) as a first priority. If there is no way to get rid of the transcoders then will be focused on minimizing the transcoding processes (TFO). Detailed information about TFO and TrFO will follow in the next sections.

The Bearer Independent Call Control (BICC) protocol supports transcoder codec negotiation capability. BICC can be used to support transcoder free operation, therefore provides smooth interworking with STM environments. Codec negotiation reminds Out Of Band Transcoder Control function at the first sight which is a prerequisite for TrFO and TrFO/TFO internetworking.

3.5.6 BICC Codec Negotiation

In the cases like OoBTC implementation, it is essential to define and determine the codecs that will be used at the edges before bearer establishment – This means user equipments (UE) are to choose the most effective and common way for themselves to establish the communication in between.

- A BICC codec negotiation starts when one of the ISNs is triggered by call and bearer SETUP request within an IAM message. As mentioned before call will be set up beforehand of the bearer. And codec negotiation capable networks – will also implement and determine the codecs also before the bearer setup.

- The originating Control function then sends a list of supported codecs, in the descending order of priority within an IAM.

- The controlling function of intermediate TSN – if exists –analysis the list of codecs and removes the unsupported codecs and afterwards passes it to the destination node.

- Destination ISN analyses the codec list and selects the most convenient codec type labeled as ‘selected codec’ along with the other available codecs in a list via APM message in the first backwards direction call control message.

- On the first backwards directed call control message –intermediate TSN node decides the codec to be used comparing and analysing the IAM connect forward and APM messages.

- Afterwards bearer is set up according to the BICC protocol on the codec and mode that have been decided by OoB codec negotiation.

The overview of the BICC codec negotiation process steps can be seen in Figure 3.10

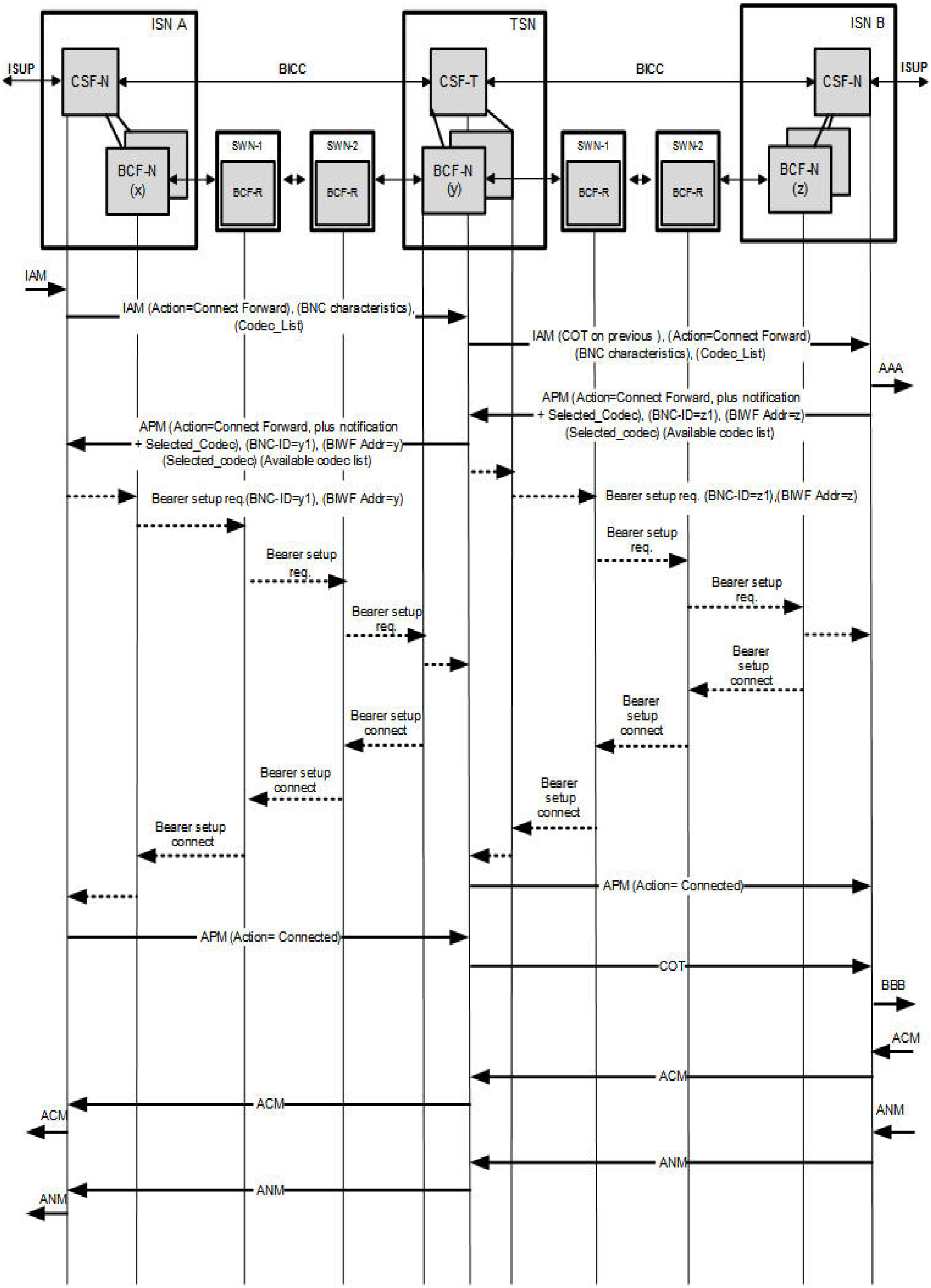

Figure 3.10 : BICC forward bearer setup with codec negotiation [5]

3.6 SIP for Telephones (SIP-T)

SIP-T is a protocol extension; defined by IETF in 2002, aiming to provide ISUP services over the SIP networks as SIP message attachments. SIP-T specification for an integrated approach allows PSTN networks connect through a packet network transmission medium. However through the researches, it is seen that ATM applications of SIP-T are uncommon, in other saying there is no SIP in ATM. Therefore the section will focus on IP networks. The biggest achievements about SIP-T are protocol translation and transparent feature exchange through the SIP and ISUP networks.

3.6.1 SIP-T Architecture and Operation

This section will focus on the functionality of SIP-T based on some possible architectural compositions. To be more specific SIP-T architecture is mainly defined by two applications; Gateway and Bridging can be seen in Figure 3.11

PSTN – IP –PSTN

This network composition is bridge functionality and the case where the SIP based network provides session control between two PSTN entities. A basic call setup procedure functions this way:

The border elements in SIP based network requires: IAM call setup, ISUP message parameter and SIP user agent client that indicates the ISUP message as an sdp attachment when forwarding the request. Afterwards in the packet network side, conventional SIP routing mechanism continues. The sdp attachment in the SIP message defines type of transport to be used for the relevant ISUP message – so to say for audio packets, sdp can associate with RTP or AVP - far end termination points, proxy server address and parameters if any available-and/or application of specified codec procedures defined in RFCs.

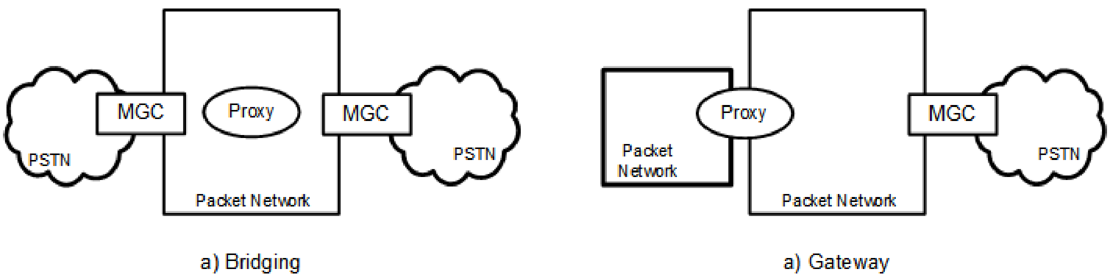

Figure 3.11 : SIP-T Bridging & Gateway

PSTN - IP

This is the scenario where one end is PSTN and far side is IP based – or vice versa - networks. It is gateway functionality of SIP-T.

In the case that the PSTN is originator side; then the procedure is that IAM from ISUP side triggers the Media Gateway Controller (MGC) to create the SIP request message attaching the IAM parameters as sdp. However if the originator side is IP (e.g. SIP Phone), there is no need to have ISUP parameters in the setup request. Accordingly the routing, side parameters and SIP encapsulation will be used through till the PTSN border element MGC unchanged. Then in MGC SIP headers will be derived in compliance for ISUP messages.

Encapsulation

One of the biggest specialties of SIP-T is encapsulation of PSTN signaling. SIP-T definition is formed by multipart MIME bodies to enable SIP messages to contain more sdp attachments. The ISUP MIME methodology enables the border elements of the networks recognize the ISUP type and determination if the variants are supported. ISUP encapsulated SIP INVITE message can be seen in Figure 3.12

```
INVITE sip:13039263142@Den1.level3.com SIP/2.0
         Via: SIP/2.0/UDP den3.level3.com
         From: sip:13034513355@den3.level3.com
         To: sip:13039263142@Den1.level3.com
         Call-ID: DEN123199902171209550099@Den1.level3.com
         CSeq: 8348 INVITE
         Contact: <sip:jpeterson@level3.com>
         Content-Length: 436
         Content-Type: multipart/mixed; boundary=unique-boundary-1
MIME-Version: 1.0

         --unique-boundary-1
         Content-Type: application/SDP; charset=ISO-10646
         v=0
         o=jpeterson 2890844526 2890842807 IN IP4 126.16.64.4
         s=SDP seminar
         c=IN IP4 MG122.level3.com
         t= 2873397496  2873404696
         m=audio 9092 RTP/AVP 0 3 4
         --unique-boundary-1
Content-Type: application/ISUP; version=nxv3;
         base=etsi121
         Content-Disposition: signal; handling=optional
01 00 49 00 00 03 02 00 07 04 10 00 33 63 21
         43 00 00 03 06 0d 03 80 90 a2 07 03 10 03 63
         53 00 10 0a 07 03 10 27 80 88 03 00 00 89 8b
         0e 95 1e 1e 1e 06 26 05 0d f5 01 06 10 04 00
         --unique-boundary-1–
```

Figure 3.12 : ISUP encapsulated SIP INVITE message

Translation

One of the other biggest achievements of SIP-T is translation specifications. There are mainly two aspects of translation:

- **ISUP_to_SIP-T message mapping:** This description denotes a mapping between ISUP and SIP at the message level. In terms of conversion; gateways are authorized with the task of reproducing the relevant ISUP message for each SIP message received (and vice versa).

- **ISUP parameter-SIP header mapping:** A SIP request that is used to set up a telephone call should contain information that enables it to be appropriately routed to its destination by proxy servers in the SIP. RFC 3372 defines the procedure for translation of information from ISUP to SIP conversion such as; the Called Party Number in an ISUP IAM must be mapped onto the SIP 'To' header field as Request-URI [20]

3.7 Session Initiation Protocol ISUP Encapsulated (SIP-I)

The session initiation protocol ISUP encapsulated has been defined by ITU-T in 2004 – after the experiments of SIP-T have occurred in need of a new discipline. SIP-I methodology by ITU-T is a very well-studied version solution to the interconnection of both ISUP and BICC versus SIP based networks. In general perspective; although SIP-I and SIP-T are almost identical in both operation and messaging skills – There are yet some essential differences between:

- SIP-I defines mapping from SIP internetworking to ISUP and BICC, while SIP-T interworks with only the ISUP.

- SIP-I includes the additional services for telecommunication interconnection.

SIP-I of ITU-T comes with a string of new achievements and definitions following;

Interworking Units (IWU): Are the border crossing elements between networks. They may be found in the networks stand-alone or with an ISUP exchange or BICC ISN nodes.

Network-to-Network interface (NNI): The interface between the IWU and ASN is named as NNI.

SIP-I Definition of SIP-to-ISUP Interworking

As the usual procedure, the IWU – which is triggered by the network for a call setup encapsulate the ISUP message into the SIP parameters executing the conversion functionality. Through the parameters in SIP-I model, as it differs from SIP-T ISUP encapsulated SIP INVITE message in Figure 3.12 should be inserted a new Content-Type parameter.

```
Content-Type: application/ISUP; version = itu-t92+;
```

It is defined in the technical specification; the conventional service requests are executed via a trusted Adjacent SIP Node (ASN) in the SIP domain.

3.8 IP Multimedia Subsystem (IMS)

The real difficulty of implementing IP-based voice and data services in the traditional networks have pushed the researches to create a mechanism that IP-enabled devices can establish peer-to-peer and peer-to-service connections easily, fast and reliably. An IP based multimedia services assembly would have really fitted this purpose.

IP multimedia Subsystem IMS is a SIP based access-independent and standard based IP connectivity and service control architecture that enables various types of multimedia services to end-users using common internet based protocols [30]

3.8.1 Main Functionalities

IMS main functionalities are following:

IP Multimedia Sessions

Main functionality of IMS is, to be able to cooperate with variety of IP based services which cannot be maintained in existing circuit-switched communication. One of the biggest advantages of IMS is that users can initiate and integrate voice, video and text services all together into for example content sharing applications and add or drop any means of services when they want.

QoS for IP Multimedia Services

IMS provides an end-to-end Quality of Service and ensures to check and apply the QoS specifications to the IP multimedia traffic. During SIP session initiation or modification phase, IMS also allows User End equipments to negotiate as Media type and direction, packet size, buffer size, RTP payload usage for media types and BW adaptation parameters in the name of QoS adjusting.

IP Policy Control

Policies in IP based networks are one of the methods to limitate, to allow, to deny or to ban some entities across the network. IMS sessions use policies to authorize and check if the signaling parameters are fully applied to the specific sessions or not.

Secure Communication

IMS introduces its unique authorization and authentication methods over the SIP security assets. However facing forehand Radio and Core networks increases the vulnerability of the security of IMS, there are some specifications to prevent such vulnerabilities for example IMS ensures that the users are fully authenticated before starting to use the services. Through the scope of the thesis security impacts will not be deeply detailed.

Access Independence

IMS is able to interwork with any means of access technology running over IP packets. This specialization makes IMS more and more flexibly comparing other Session Control Protocols and making it even more desirable.

3.8.2 Architecture

IMS as a multifunctional service control architecture having multiple means of entities. Those can be classified as following:

Session Managements and Routing Family: CSCFs
Databases: HSS, SLF
Services: Application Server, MRFC, MRFP
Interworking Functions: BGCF, MGCF, IMS-MGW and SGW
Support Functions: PCRF, SEG, THIG
Charging

As IMS has really many intermediate equipments but this chapter will outline the core routing nodes to supply a better understanding. A generic IMS architecture illustration can be seen in Figure 3.13

Proxy - Call Session Control Functionality (P-CSCF)

P-CSCF is the first contact point for users within the IMS. On other saying all the traffic from User End equipment will be sent to P-CSCF first. UE equipment gets the IP address of P-CSCF at the first contact to the network otherwise communication cannot be done. Some main tasks are following:

SIP compression
Border gate Equipment behavior to the UE and between trusted/untrusted networks
IPSEC security association
Emergency Session Detection
Interaction with Policy Decision Function
Generates Charging Records

Interrogating - Call Session Control Functionality (I-CSCF)

I-CSCF is the interface point to an operator's network for all the connections belonging to the subscriber of the network operator. I-CSCFs like other CSCFs ,can handle more than one IP address and these addresses are published via DNS so that SIP request from another domain can find the relevant I-CSCF.

Some main tasks are following:

Obtaining the name of the next node that serving the user, from HSS via diameter query
Always located at the Home domain of the user
Routing the incoming SIP requests to a relevant S-CSCF
Assigning S-CSCFs for the specific users

Serving - Call Session Control Functionality (S-CSCF)

Serving- CSCF is mainly responsible for registration of IMS users after downloading the service profile of the user from HSS.

Some main tasks are following:

Notifying P-CSCF for each registration event
Responsible of correct routing of session requests to the destination
Enforcing operator policies when required
S-CSCF acts like a notifier for Registration event informing AS

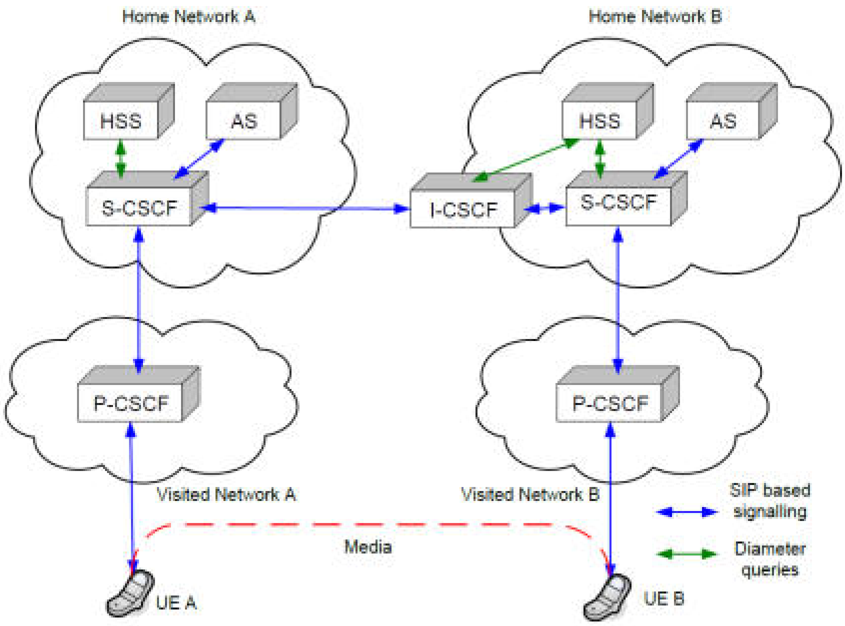

Figure 3.13 : Basic IMS Core Architecture – Roaming [29]

3.8.3 Multimedia Handling

Multimedia handling of IMS based services uses the call session control functions to maintain the route control-plane signaling between UEs. The scope of this section is to explain the media path for different kinds of traffic. An illustration of such scenario can be seen in Figure 3.14.

IMS supports simultaneous transfer of single/multiple media components with real-time specifications. Voice, Video and Text media components are taken as core components in multimedia communication. As a common specification those components are transported between UEs using Real-Time Transport Protocol (RTP).

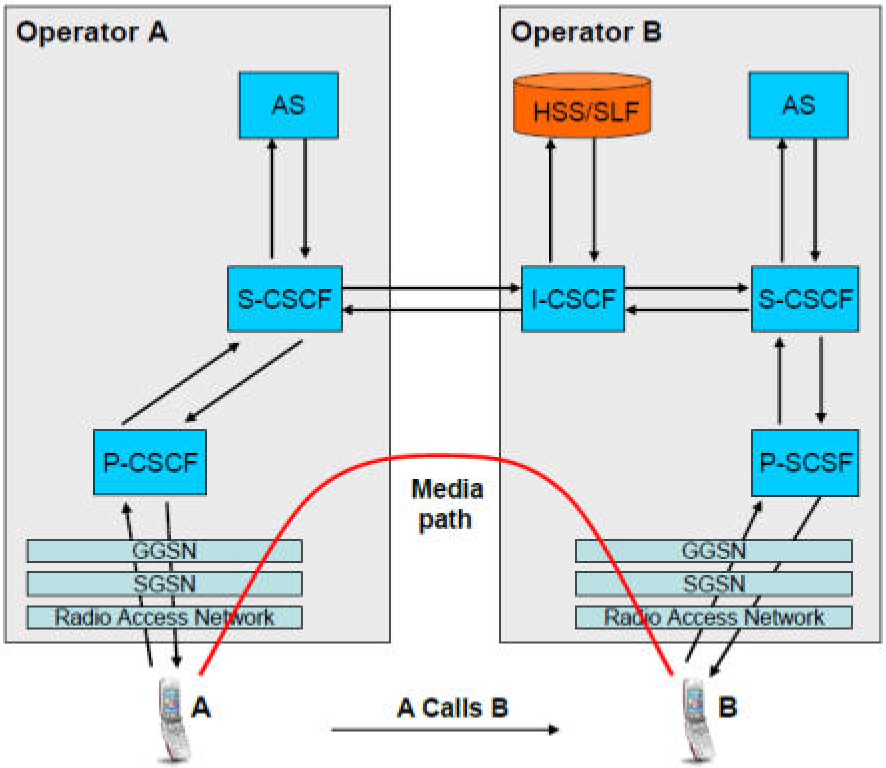

Figure 3.14 : IMS UE-to-UE call illustration

Recommended Codecs for Voice in Mobile Networks:

AMR-NB speech codec including all 8 modes and source controlled rate operation.

AMR-WB codec @ 16 kHz sampling frequency; including all 9 modes and source controlled rate operation.

Recommended Codecs for Video in Mobile Networks:

H.264/ MPEG-4

H.263

Recommended Codecs for Text in Mobile Networks:

T.140

Session Setup

In the session setup phase essentially matching at least one RTP profile to RTP media stream; RTP shall for each media determine:

IP Addresses
RTP Profile (Exchanging SDPCapNeg parameters within SDP Offers)
UDP Port Numbers
Codecs
RTP Payload number
RTP Payload Formats

Bandwidth Negotiation

SDP includes the bandwidth information for each media stream and for the whole session.

4 Codecs in Networks

4.1 Introduction to Codecs

As it is already studied in Section 2.3, in the mobile networks, voice and data calls are allocated to traffic channels (TCH), with usually 8 channels per transceiver. In frequency planning, a subset of all available carriers is allocated to TCHs and each transceiver is associated with a carrier frequency. Nevertheless frequency band in the radio interface between a mobile user terminal and a base station is limited. But there are several ways to use assigned frequencies decisively named: Network Dimensioning.

Mobile network dimensioning is a set of different mechanisms and is indeed essential and the upmost requirement that available resources should best accommodate the service and capacity requirements of the user number. That means networks tend to supply their services to the users as well as possible within the limited air interface sources. Significantly, the efficient use of bandwidth in the mobile systems is done by re-forming (compressing/decompressing) the speech signal. These re-forming mechanisms are known as codecs. Compression techniques are a clever way to keep resources in optimum level by saving money in that interface. A typical full rate channel (16kbit/s) utilizes a compression rate of 4:1. A half rate channel (8kbit/s) is half of that and it operates at compression rate of 8:1. The G.711 standard is a 64kbit/s common reference point for "real" speech codecs.

The 64kbit/s phenomenon comes from a very basic calculation. Starting with the frequency range for the human speech for the auditory system happens to be between 300-3400 Hz. To ease the calculation the upper limit 3400 Hz is taken and rounded to 4000 Hz which is 4 KHz. Due to the sampling theorem; in order to reproduce the original signal after sampling, a sampling rate that is double the desired frequency band must be used. Accordingly speech in telecommunication networks is sampled at 8 kHz. If the sampling rate is less than double the signal cannot be reproduced. As it can also be seen in Section 2.1.3 PCM sampling uses 8 bits for quantization at 8 KHz which makes a 64 Kbit/s native, plain speech stream.

However lossy compression has always some effects on speech quality and more compression occur in severe signal distortions, losses and discrepancies. That means more compression results in less quality of speech signals. Systems have to be introduced with the best suiting codecs to keep and empower the acoustic-phonetic information of the signal sent over the air interface.

Codecs are the algorithms which used to convert the analog audio signal to digitally encoded version. There are simply a lot of codecs are introduced for different purposes. Those different codec types differentiate in the outcome quality, the bandwidth requirements and compatibility with the services. It should be noted that each intermediate node in the network, service, phone or gateway generally supports several different codecs. Speech codecs are typically designed to maximize a mean opinion score (MOS) of the codecs. MOS values are perceived quality of audio after compressed by the particular codec, transmitted, and decompressed. The score is assigned objectively by a group of listeners using the procedure specified in ITU-T standards P.800 and P.830. MOS values can be seen in Table 1 [31]

5	Excellent
4	Good
3	Fair
2	Poor
1	Bad

Table 1 : MOS values

4.2 Overview of Mostly Used Speech Codecs

4.2.1 G.711 (PCM)

G.711 is a high bit rate (64 Kbps) ITU standard codec. It is the native transformation technique in networks. Importance using here the word ‘transformation’ is because PCM is not a compression technique. It is just simple transformation of analogue signal to digital format. That’s why the voice quality is really high (MOS value of 4.2) and the latency is very low in the network as there is no compression/decompression process for G.711 signal.

However the only disadvantage of G.711, the bandwidth utilization is really high and can sometimes be up to 84 Kbit/s when there are additional TCP/IP overheads. In this example a 128 kbit/s channel is needed on both side of the communication which indeed can be very problematic and expensive when it comes to deploy such infrastructure.

There are two versions: A-law and µ-law. U-law corresponds to the T1 standard used in North America and Japan. The A-law is of the E1 standard used in the rest of the world (see Figure 1.4). The biggest difference is the sampling method of the analog signal. In both schemes, the signal is not sampled linearly, but in a logarithmic fashion. A-law provides more dynamic range as opposed to µ -law and this result in a less blurry sound.

4.2.2 Adaptive Multi-Rate Codec Type (AMR)

The native AMR codec family also known as AMR Narrow Band (AMR-NB) was introduced by the 3GPP Forum for GSM and UMTS technologies under the technical specifications of TS 26.090 and TS 26.093. The significance of AMR speech codecs is being adaptive, offering a set of speech rates that can be provided on different network conditions. Dynamic rate assignment mechanism is only possible when the channel is assigned for a voice conversation, enabling the AMR speech codec throughout the call, based on current channel conditions. AMR provides signals at different modes with variable bit rates from 4.75 to 12.2 kbit/s.

Multiple speech codec rates and dynamic rate switching is very useful to maintain the voice quality during unpleasant radio frequency and channel conditions. If the channel conditions are good then the half-rate (HR) channel can be used with AMR codecs that means the call capacity is double comparing full-rate (FR) channels. However, if channel conditions get worse, codec rates are dynamically modified to accommodate FR channel conditions after the decision mechanism on the most appropriate codec mode to apply at a given time. Codec mode adaptation for AMR is based on received channel quality estimation in both MS and BTS. AMR-NB codec modes can be seen in Table 2 [32]

AMR Codec Mode	Bit rate	Channel Type
AMR_4.75	4.75 kbit/s	FR / HR
AMR_5.15	5.15 kbit/s	FR / HR
AMR_5.9	5.90 kbit/s	FR / HR
AMR_6.7	6.70 kbit/s	FR / HR
AMR_7.4	7.40 kbit/s	FR / HR
AMR_7.95	7.95 kbit/s	FR / HR
AMR_10.2	10.20 kbit/s	FR
AMR_12.2	12.20 kbit/s	FR
AMR_SID	1.80 kbit/s	FR / HR

Table 2 : AMR-NB codec modes

4.2.3 Adaptive Multi-Rate Wideband Codec Type (AMR-WB)

The WB-AMR speech codec family is standardized by 3GPP TS 26.190 and ITU-T G.722.2. The introduction of AMR-WB is significant because the first time that the same codec has been adopted for both wireless and wire-line telecommunications [32].This approach has the advantage of being able to bypass transcoding and therefore ease the implementation and integration of more speech and audio applications and services across the various communication systems and platforms.

Wideband AMR (AMR-WB) codecs uses an audio bandwidth of 50 Hz to 7 kHz and sampling frequency up to 16 KHz which provides a much wider spectrum of sound than previous methods. The codec is suitable for phone conferences and can transmit music in an acceptable quality. As in the case of AMR-NB, several data rates between 6.60 kbit/s and 23.85 kbit/s are specified with AMR-WB. AMR-WB is based on the same principle as AMR-NB; as the connection quality becomes poorer, the data rate decreases and error protection increases. As a result, AMR-WB provides speech quality better than the experienced in narrow-band wire-line telephone and wireless networks. AMR-WB codec modes can be seen in Table 3

AMR Codec Mode	Bit rate	Channel Type
AMR-WB_1.75	1.75 kbit/s	FR / HR
AMR-WB_6.60	6.60 kbit/s	FR / HR
AMR-WB_8.85	8.85 kbit/s	FR / HR
AMR-WB_12.65	12.65 kbit/s	FR / HR
AMR-WB_14.25	14.25 kbit/s	FR / HR
AMR-WB_15.85	15.85 kbit/s	FR / HR
AMR-WB_18.25	18.25 kbit/s	FR / HR
AMR-WB_19.85	19.85 kbit/s	FR / HR
AMR-WB_23.05	23.05 kbit/s	FR / HR
AMR-WB_23.85	23.85 kbit/s	FR / HR

Table 3 : AMR-WB codec modes

4.2.1 GSM Full Rate Codec Type (GSM-FR)

This form of voice codec was the first speech codec used with GSM and it is introduced after tests were undertaken to compare it with other codec schemes of the day. The speech codec is based upon the regular pulse excitation and baseband coding mechanisms. The performance is limited by the tonal noise produced by the system.

4.2.2 GSM Enhanced Full Rate Codec Type (GSM-EFR)

GSM Enhanced Full Rate (GSM-EFR) (or GSM 06.60) is a speech coding standard that is defined in order to improve the poor quality of GSM-Full Rate (FR) codec. GSM-EFR codec is compatible with the highest AMR modes and operates at 12.2 kbit/s and provides wire like quality in any noise free and background noise conditions. Enhanced Full Rate helps to improve call quality on the other hand this codec have higher computational complexity, which in a mobile device can potentially result in increase of higher energy consumption.

4.2.3 GSM Half Rate Codec Type (GSM-HR)

This GSM standard allows the splitting of a single full rate voice channel into two sub- half rate channels doubling the number of voice calls that can be handled by the network with very little additional investment.

The GSM Half Rate codec uses the codec algorithm that codes the data around 20 ms frames each carrying 112 bits to give a data rate of 5.6 kbps. This includes a 100 bps data rate for a mode indicator which details whether the system believes the frames contain voice data or not. This allows the speech codec to operate in a manner that provides the optimum quality. The Half Rate codec system was introduced in the 1990s, but in view of the perceived poorer quality in compare with full-rate codecs, it was not widely used.

5 Tandem Free Operations (TFO) and Transcoder Free Operations (TrFO)

5.1 Introduction

In mobile networks, speech signals always involve several transcoding steps. As already studied, compression is essential in the radio network in order to transmit the speech signal through the core network over the standard G.711 PCM codec over 64 kbit/s circuit switched links. On the other hand, quantized speech data stream to G.711 codec is essential for fixed line-to-mobile calls. An advantage of voice transcoding in the network is that data stream is protected from noise and echo addition in the speech path.

The introduction of the Bearer Independent Core Network functionality in the core network, most of the connections are based on high bandwidth ATM or IP links rather than TDM links. In such networks it is therefore possible to transmit a voice data stream in the core network with other codecs such as AMR-WB codecs than G.711. This also has the additional benefit of removing transcoders from the speech path which reduces delay and improves speech quality as the speech signal is no longer degraded by the transcoding process.

This flexibility of removing transcoders has widened the vision through a call session that no transcoders are involved – which will be named after Transcoder Free Operations (TrFO). In the situations that transcoders must be used in the network, it was clear that at least tandem transcoding processes can be bypassed – which will be named after Tandem Free Operations.

The details about these two brand new techniques are following:

5.2 Tandem Free Operations (TFO)

5.2.1 TFO Overview

Although TrFO is a call priority within the networks it cannot be implemented in every case. For the situations where more than one TC is needed the TFO option is considered. This chapter includes detailed operation of TFO. Figure 5.1 shows TFO transcoder arrangement

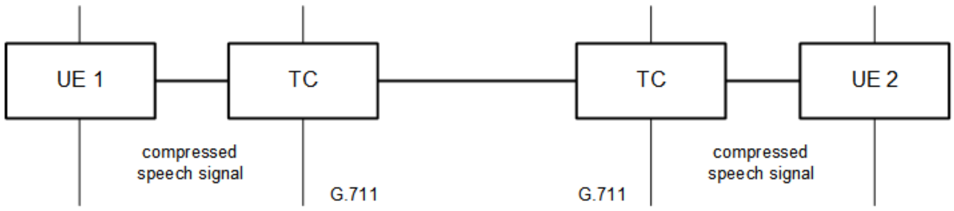

Figure 5.1 : TFO Transcoder arrangement

The first overview of Tandem Free Operation begins with an accurate process overview in GSM and/or UMTS networks as Tandem Free Operation is applicable in both GSM and UMTS networks.

In the traditional PLMN networks; based on GSM TRAU, BSC and in UMTS MGW and MSC equipments are operating encoding/decoding in tandem (multiple sequences) resulting voice quality degradation. The degradation is even getting worse when the speech signals are used of low frequencies. Thinking over -forcing the solution to eliminate this problem by removing the sequential transcoding operations in the voice path only under the circumstance that if the two MS/UEs are using the same codec. TFO intends to avoid the traditional intermediate double speech/data signal transcoding in the calls leaving the transcoding only to the end terminals. TFO is an in-band codec negotiation protocol. Unlike TrFO – codec negotiation can only be done only after the call setup. Call setup procedure in TFO mechanism is identical to BICC.

In general perspective TFO functions this way;

Without any trigger of call _SETUP, both end transcoder units do frequently communicate with each other over 64 Kbps PCM samples – exchanging also TFO messages replacing[4] the LSB of every 16th speech sample - with procedure parameters attached - in terms of readiness, supported codecs and compatibility for additional operations such as TFO. In case it is understood by both TCs that they have same speech codec types and configurations - therefore compatible for TFO, transcoders activate TFO

[4]This method is also called PCM Bit stealing

operation automatically. After the call setup then TCs distinguish the full establishment of TFO protocol.

5.2.2 TFO Frame Structure & Messaging

TFO Frame Structure

Frame formats in TFO discipline, are pretty much depending on the Codec_Types parameter that has been agreed by both of the transcoder equipments. The two transcoder units in a TFO call have a connection of 64 Kbps link with an average 8000 Hz sampling frequency. During the communication process between the local and remote transcoders, via PCM sample, TFO frame information is placed into the two least significant bits of each 8-bit octet of the 64 Kbps link.

A TFO frame contains N times 8-bit octets and gathers a period of 20 ms. PCM coding takes place in most significant (MSB) 6 bits and the rest Least Significant (LSB) 2 bits are for compressed speech signal, TFO frames and control bits. An illustration to TFO frame can be seen on Figure 5.2

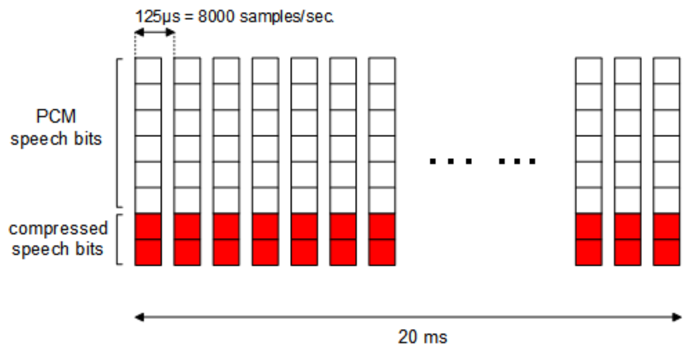

Figure 5.2 : TFO Frame Structure

TFO frames are including information parameters such as bits for synchronization, bits delegated to be used in codec operations, Cycling Redundancy Check (CRC) parameter and bits for phase adjustment.

Transcoding functions then need to sub-multiplex the generic frames into frequency varied formats – in GSM between A and Ater interfaces and in UMTS between IuCS and Mc interfaces. TFO frame formats for 8 Kbps, 16 Kbps, and 32 Kbps sub-multiplexing processes are defined for following Codec_Types:

- GSM full rate (GSM-FR) – 16 Kbps
- GSM half rate (GSM-HR)- 8 Kbps
- GSM enhanced full rate (GSM-EFR)- 16 Kbps
- AMR Family (FR_AMR, HR_AMR, UMTS_AMR…)
- AMR-WB Family (UMTS_AMR-WB, FR_AMR-WB…)

5.2.3 Inband signaling and TFO Messages

Inband Signaling (IS) messages are designed to accommodate some specifications throughout the TFO. The unique structure and bit placing prevent them to eliminate the IPEs without delay and queuing. The mechanism works mainly that IS messages replace some bits of LSB section of the PCM frame. IS messages are being transferred in the LSB section of PCM samples in the channels replacing one bit in LSB of every 16th bit of PCM sample. This structure can be seen in Figure 5.2

All of the generic IS messages includes an IS_Header parameter with the Command block next to it. Also some bits are spare to support further services and configurations. An example illustration can be seen of an IS message with two extension parts can be seen in figure 5.3

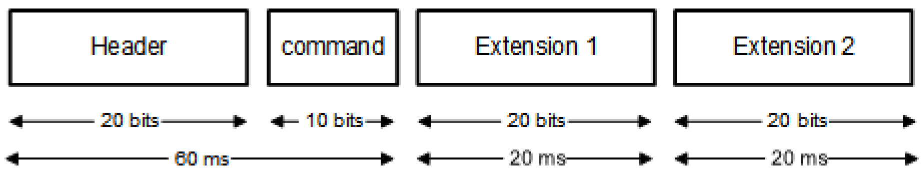

Figure 5.3 : TFO inband signaling message structure

TFO Request Message (TFO_REQ):

This message is taken as the very beginning of the TFO process. Local TC sends a string of parameters attached to the request signal, to the distance TFO capable equipment using a specific Codec_Type. Some of the parameters are:

Originator System_Identification
Originator Local_Signature
Originator Local_Used_Type
Codec_List

TFO Acknowledgement Message (TFO_ACK):

Every request message should be followed by an acknowledgement in order to confirm that the request has been received. TFO_ACK includes the same parameters as TFO_REQ but of distant side values and the second change is that Originator Local_Signature is then replaced by distant side Local_Signature.

TFO_REQ_L Message:

Request message in terms of a refreshment of TFO_REQ messages, is used for new negotiation for codec mismatch or rarely information updates. Parameters are identical to TFO_REQ message besides the Local_Codec sublist of Alternative_Codec_Types list.

TFO_ACK_L Message:

Confirmation response to TFO_REQ_L messages. Parameters and the functions are almost identical to TFO_REQ_L messages except Originator Local_Signature is replaced by distant side Local_Signature.

TFO_DUP Message:

This message type is to inform the partner TC that whilst the PCM samples are still coming, TFO messages are also received.

TFO Transparent Mode Message (TFO_TRANS):

This message is used in order to stimulate the IPEs to be ready for the TFO communication so that IPEs let the TFO frames to be transmitted transparently inside of LSBs. TFO_TRANS message includes the parameter of Local_Channel_Type (8/16/32 Kbps)

TFO Normal Mode Message (TFO_NORMAL):

This message is used in order to terminate transparent operation of IPEs and to get them back to the normal means of operation. This message type includes no parameters.

TFO Synchronization Lost Message (TFO_SYL):

Used when the sender TC fails to receive any TFO frames from the distant TC. This is a pre-warning before the synchronization is lost and TFO is released.

5.2.4 Technical Detail: Architecture, Operation & Implementations

Through the scope of the work – TFO operation will be followed via UMTS networks but although the architecture differs; TFO protocol and frame formats are used commonly in PLMN networks too. In this section TFO implementation aspects with technical details will be held.

Figure 5.4 illustrates the GSM architecture of Tandem Free Operations already established and the in TFO perspective GSM functions are following;

Ater Interface: An interface between BSC and BTS is only accommodated for AMR or AMR-WB TFO specifications. Ater aids TFO to exchange Time_Alignment and Rate_Control parameters via config frames.

Abis Interface: The interface between BSC and TRAU has the complementary function of Ater interface detailed above and as an addition it may be used with FR_AMR, HR_AMR, GSM_FR, GSM_EFR, OHR_AMR, GSM_HR, FR_AMR-WB, UMTS_AMR-WB, OFR_AMR-WB and OHR_AMR-WB codec types for the exchange of the codec configuration information.

Um Interface: The interface between MS and BSC used for Layer 3 signaling modification of codec configurations and Codec_Type parameters.

A interface: Interface between BSS and MSC - The real path of TFO communication where a direct transparent path is established.

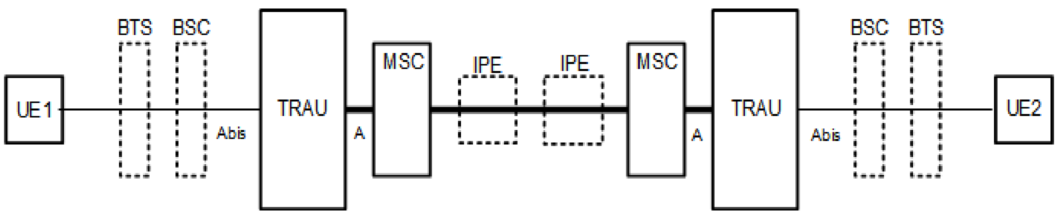

Figure 5.4 : TFO GSM Architecture

In UMTS perspective; inband signaling is between Nb interfaces of MGWs – which includes transcoding functionalities. Core networks in UMTS are transparent for speech and inband signaling. Brief comments about some interfaces are following:

E Interface: Connection between two MSCs in an UTRAN network. The best suitor for this interface is indeed ISUP - but ISUP is not accommodated in TFO. Although there are some specifications about OoBTC TFO disciplinary operation, through the complexity and current impossibilities 3GPP has chosen not to focus on it. Therefore by configuring as “No Change” MSCs are ought to be taken out of TFO scope.

Radio Access Network Application Part (RANAP): The interface between MSC and RNC is invisible in terms of TFO communication specifications.

Iu Interface: Interface between MGW and RNC – is not accommodated by TFO.

H.248: Is an interface for MGW controlling which takes essential role in execution of transcoding operations. H.248 protocol communicates with transcoding mechanism supplying Local_Codec_Type parameter and configuration. In this situation Local_Codec_List is optional.

Nb Interface: The actual TFO path is on two Nb interfaces between MGWs. Nb interface transmits the PCM samples in which TFO messages and frames are carried. TFO UMTS architecture can be seen in Figure 5.5

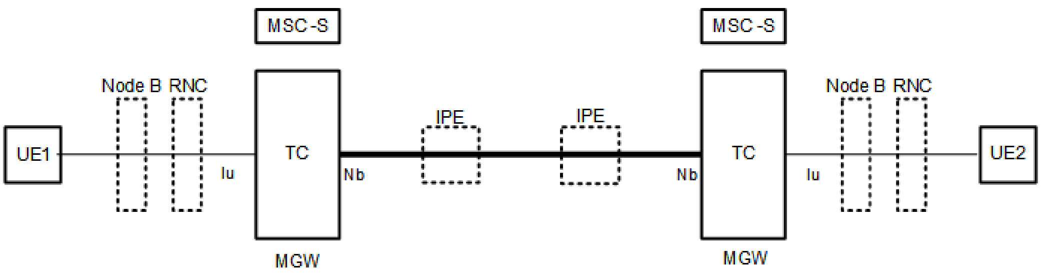

Figure 5.5 : TFO UMTS Architecture

Tandem Free Operation activation and management procedures are held by transcoder units[5]. Establishment of the communication path, arranging and managing all the equipments in between for a smooth transparent transmission and TFO termination procedures are all handled in transcoders. This is another way of proving that the transcoders are inevitable for TFO operations. Right after the TFO is activated in network – all the procedures executed are called inband signaling.

[5]All the Equipments that are on duty with transcoding. TRAU and/or MGW TC

Step-by-Step operational details are following:

Pre-Synchronization of IPEs

Shortly after TFO is enabled in the network and speech samples are exchanged between local and distant transcoders – they simultaneously start the negotiation process exchanging TFO_FILL parameter in order to keep existing/candidate IPEs synchronized and make ready for a transparent communication path establishment.

IPE modes should be configured to either transparent communication or normal. The biggest difference between IPE transparent (command: Go transparent) and normal mode (command: Go to normal) is that in transparent mode IPE output messages contain TFO awareness bits. There is also a sub-parameter in TFO speech frames called: Keep_Open. When Keep_Open parameter is detected by IPEs – it is understand that the TFO session established before still goes on. In terms IPEs do not detect Keep_Open parameter in the data stream for the threshold time 1 sec or Go to normal command is received – Transparent communication ends.

TFO Codec Negotiation

The next step of having an agreement on the configuration basics is that; the transcoders in both ends send TFO_REQ parameter to exchange system identification, Activated Codec List (ACL) and configuration details with each other. TFO_REQ is in both transcoders followed by a TFO_ACK message including system cluster identification (GSM, UMTS...), actual codec, common codec list, selected codec list, a random value for loopback detection and notifications for transcoder and MSCs if TCs have TFO capability. In case no common codec is found – procedure for codec mismatch resolution applies.

Codec Mismatch Resolution

First of all transcoders have to support codec mismatch resolution specification. It is some kind of refreshed negotiation phase but with different introduced message type. In case of codec mismatch – TCs propagate TFO_REQ_L messages supplying full list of Supported Codec_Types list for each other. This message has to be followed by TFO_ACK_L message in order for an approval. In case of a constant disagreement on the codec negotiation then transcoders no longer insist on TFO.

TFO Establishment

Following the negotiation phase, real meaning of TFO communication happens during the establishment process. Transcoders sends TFO_TRANS messages to the IPEs in order to indicate that TFO messages will be sent afterwards.

TFO will be counted successfully establish – when two transcoders have successfully deliver TFO messages encoded by the common speech codec type.

Codec Optimization

After the TFO has been established, some complementary procedures like handover procedure may result codec optimization in TFO. Supported Codec_Types list is handed over to the transcoders by TFO_REQ_L followed by TFO_ACK_L parameters like Codec Mismatch Resolution implementation. The only difference here is Codec Optimization is done inband therefore faster after TFO establishment.

When Codec Optimization result in a new Common Speech Codec – then transcoders will switch to the new codec and TFO communication will continue. Only in some cases codec switching may break TFO communication.

TFO Termination

TFO termination can be terminated when MS/UEs end the communication, different service participation, an unsuccessful handover has been done, or one or more TCs lose TFO capabilities.

Transcoder then give up TFO framing and Keep_Open frame include into the data stream afterwards IPEs will also be back to normal operation mode and TFO literally ends.

5.3 Transcoder Free Operations (TrFO)

5.3.1 TrFO Overview

Transcoder Free Operation (TrFO) is a standard mechanism to provide connection without transcoder for UMTS Bearer Independent Core Network speech calls. Its functionality is mainly defined in 3GPP TS 23.153 out of band transcoder control, Stage 2.

Transcoder Free Operation (TrFO) is a capability of the Legacy MS domain maintaining the transport of compressed speech in a packet transport network through the elimination of unnecessary coding and decoding of the signal by intermediate elements in the bearer path. The operation is based on codec negotiation and selection performed by the MSC Server and the user plane operations of the MGW, including user plane protocol handling and automatic transcoder removal and insertion, when applicable. Transcoding functions such as TRAU and TC may also be associated when needed. By transporting only the compressed speech, TrFO achieves bandwidth efficiencies in user plane (bearers) and reduces delays occurred by unnecessary transcoding and also increases voice quality – aiming as high speech quality as possible with as low bandwidth usage as possible. The normal and desired scenarios can be seen in Figure 5.6 and 5.7

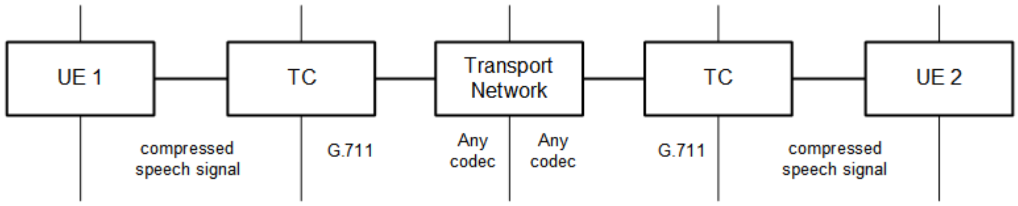

Figure 5.6 : Typical transcoder arrangement

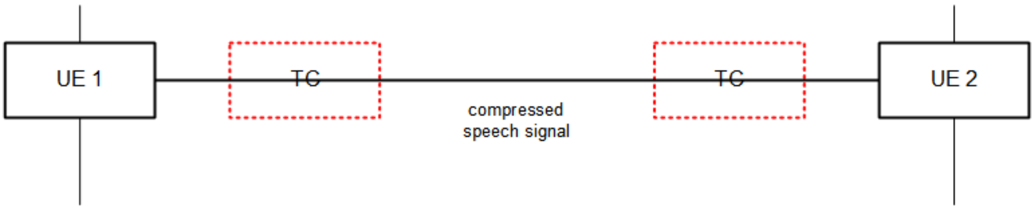

Figure 5.7 : TrFO transcoder arrangement

Transcoder Free Operation discipline does codec negotiation on a higher protocol layer by out-of-band signaling. The most important difference between TFO and TrFO is that in case of TrFO the whole transport network control layer must be fully aware of the implications of the coded speech that is transported [16]. Additions of supplementary services also have to work that way. Typically MGWs are responsible to take care that the correct coding is applied. TrFO utilizes out of band signaling capabilities that include the ability to determine the negotiated codec type to be used at the two end nodes. If the two end nodes are capable of the same codec operations, it may be possible to traverse the entire packet network using only the compression of the preferred codec [17]

The big advantage of TrFO is: the coded traffic stream is not interrupted without careful consideration of the consequences. Transitions can be controlled in a better way.

Requirements

The TrFO network operation therefore has some requirements to function. Packet based networks may enable TrFO capability fully only in mobile-to-mobile communication at the circumstance that AMR Codec_Types and Codec_Attributes parameters match exactly in both ends with User Plane 2 supporting Iu and Nb interfaces. Additionally TrFO is applicable to WCDMA (with ATM or IP bearer technologies) and to GSM (when AoIP is used). In Mobile-to-PSTN calls where only a single transcoder is needed between BICN to PTSN boundary – then the protocol is named as Remote Transcoder Operation (RTO).

TrFO supports:

- UMTS_AMR2 (Set 7) – single mod codec with 12.2 Kbps
- UMTS_AMR2 (Set 1) – includes 12.2, 7.4, 5.9 and 4.75 Kbps modes
- GSM FR
- GSM HR
- EFR
- FR AMR (Set 1)
- HR AMR (Set 1)
- FR _AMR-WB (Set 0)
- UMTS_AMR-WB (Set 0)

Normal Procedures

The TrFO is reported revertible; means if through any circumstances a change in the call state happens – such as no more UE signal compression is available, one or both users are roaming through the other types of networks that do not support TrFO or codec modification is needed – and TrFO is not available, communication continues in TFO or traditional way. Hence when the conditions are suitable again for the TrFO communication – the call shall continue over TrFO. This is done by MGWs inserting a TrFO BREAK function in order to use previously stored RFCI values so that it can continue with the Iu framing protocol functions. A normal TrFO call setup flow can be seen on Figure 5.8

Successful Outcome

TrFO is said to be successfully integrated – when the data stream trespass the whole packet network with only the speech compression for air interface in UEs.[6]

[6] Compression of data stream on UE is inevitable as Analog-to-Digital conversion of the stream must be done at least once.

User Involvement

TrFO as a mechanism from beginning to end is decided, controlled and executed by BICNs. End users are not aware of the negotiation, call and bearer setups in core network as no authorization, de- authorization, registration, de- registration and activation is needed and/or required from user side.

5.3.2 Technical Detail: Wireless System TrFO Architecture& Function

In this section the technical aspects of TrFO will be given in details supporting with call flows on examples.

Out of band call SETUP

Call SETUP in TrFO starts with originating user end equipment (O-UE) sending the request through IAM messaging to its relevant Originating MSC (O-MSC)[7]. Originating MSC then attaches the available Codec_List parameter passes it to terminating MSC (T-MSC) using BICC path through the network (Section 3.5). It should be noted that all intermediate MGWs – if available – will also detect, analyses and edit the available Codec_List parameter if there is any codecs and/or modes that are not supported along the path. In the worst case PCM G.711 is the base codec to use in communication. Terminating MSC then checks terminating user end equipment (T-UE) and makes the decision for the codec – attaching Selected_Codec parameter to the IAM message along with its own available Codec_List. Each MSC here can define its Codec_List and the priority of the codecs by their own. A plain TrFO call setup flow can be seen in Figure 5.10. Codec_Lists are formed by Codec_Types and Codec_Attributes parameters. Detailed information of codec lists and where they are ordered can be seen in table 4

Supported Codec List (DTAP)	List of codecs supported by end user equipments
Supported Codec List (BSSMAP)	List of codecs supported by BSS-SCLs
Supported Codec List (BICC)	Ordered list used in NNI OoBTC signaling
Available Codec List (BICC)	List of codecs available for NNI connection
Selected Codec (BICC)	List of codecs selected to be used on NNI connection

[7]It has to be stressed that an important assumption here is that each MSC has a static knowledge about codec capabilities of its MGW

Iu-Supported Codecs List (MAP)	Ordered list used for MAP signaling from O-MSC to T-MSC
Iu-Available Codecs List (MAP)	List of codecs available for target Iu interface
Iu-Selected Codec (MAP)	List of codecs selected for target Iu interface
Iu-Currently Used Codec (MAP)	Codec in use on serving Iu interface prior to inter-MSC handover
TFO Codec List (H.248)	List of codecs that MGW used for TFO in-band negotiations
Distant Codec List (H.248)	Codecs MGW get from distant node for TFO in-band neg.
Codec (H.248)	To use at a certain MGW termination
MSC Preferred Codec List (BSSMAP)	The list of codecs supported by both MSC and end user eq.
AoIP-Supported Codecs (Originating)List(MAP)	List of supported codecs from O-MSC to T-MSC in case of AoIP
AoIP-Selected Codecs (Target) List (MAP)	List of codecs that selected by T-BSS
AoIP-Available Codecs List (MAP)	List of codecs for GERAN A/Gb mode for target AoIP

Table 4 : Overall OoBTC function codecs and lists

In this stage TrFO can be established only if the returned codec list includes any common codecs other than PCM G.711. On the other hand if no common codecs can be found except for PCM G.711 then the connection is not possible as there is no fall back mechanism in TrFO – call has to be established from the beginning on with transcoders are involved to use another method.

Inband signaling in TrFO

The next step after the call and bearer has been setup, in TrFO in general is being referred as inband. Including– end to end inband signaling of TrFO mechanism manages Radio Bearers (RAB) initialization, bad frame handling, packet synchronization, RTO time alignment, SCR/DTX handling and specifically for AMR codecs ACS subset definition and Rate control functionalities. Detailed information about the functionalities will follow in the next chapter.

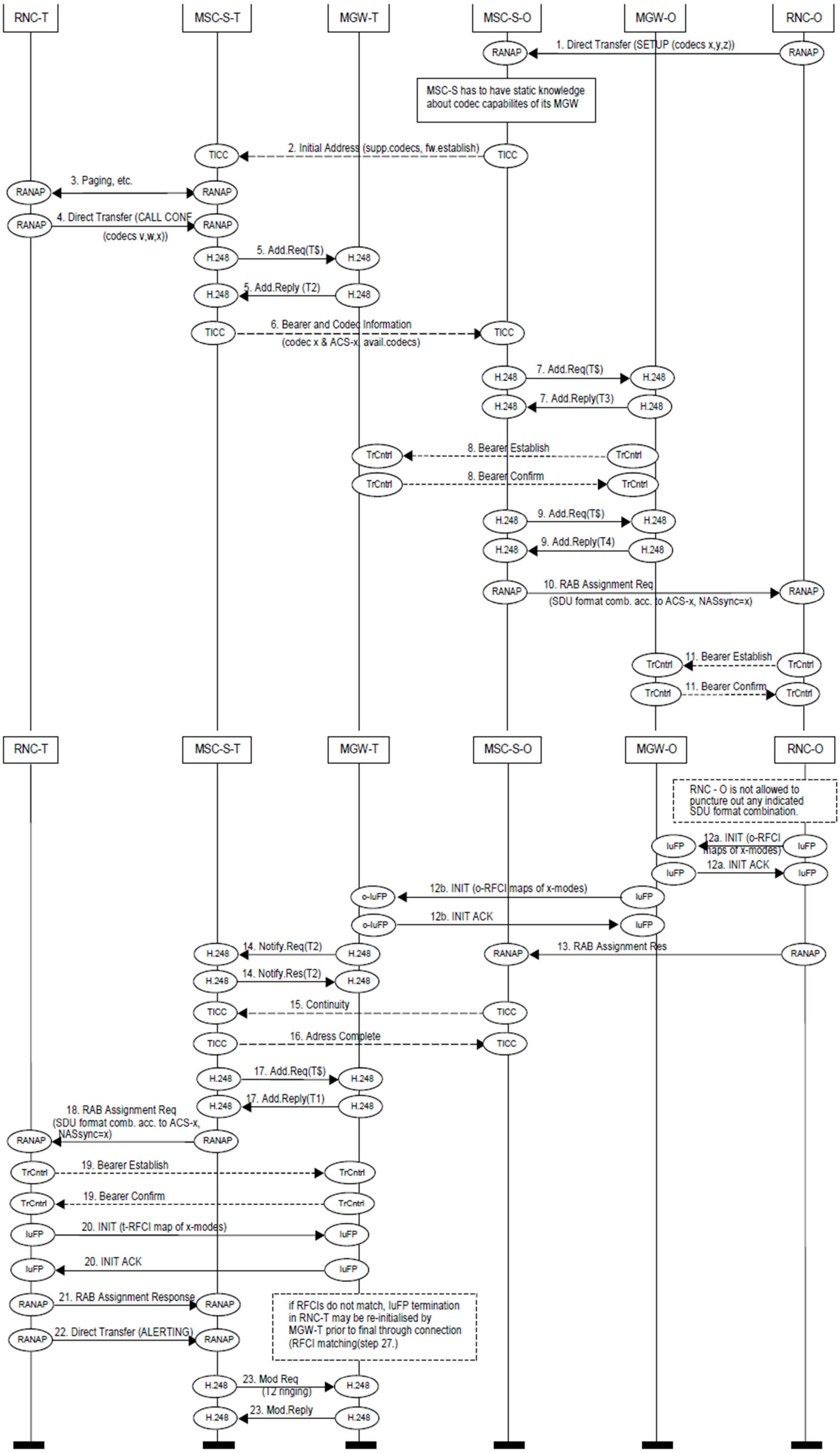

Figure 5.8 : TrFO mobile-to-mobile call setup flow [1]

5.3.3 TrFO Mobile-to-Mobile Communication

This section will cover mobile-to-mobile communication and node functionalities for TrFO mechanism. It should be noted that TrFO mechanism differs in GSM and UMTS architecture as the nodes and the procedures having differences therefore both architecture will be mentioned in this section. However the main focus will be UMTS network and TrFO UMTS functioning. TrFO UMTS-to-UMTS and UMTS-to-GSM architectures can be seen in Figure 5.9 and 5.10

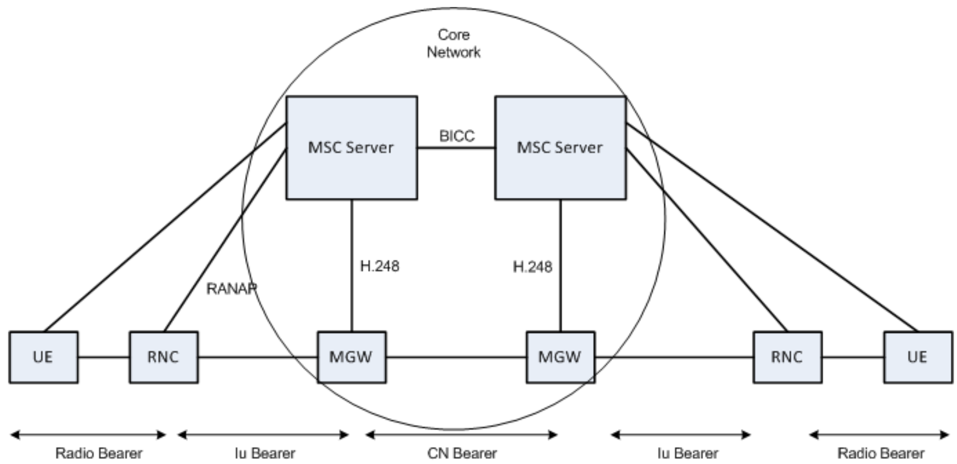

Figure 5.9 : TrFO UMTS-to-UMTS architecture

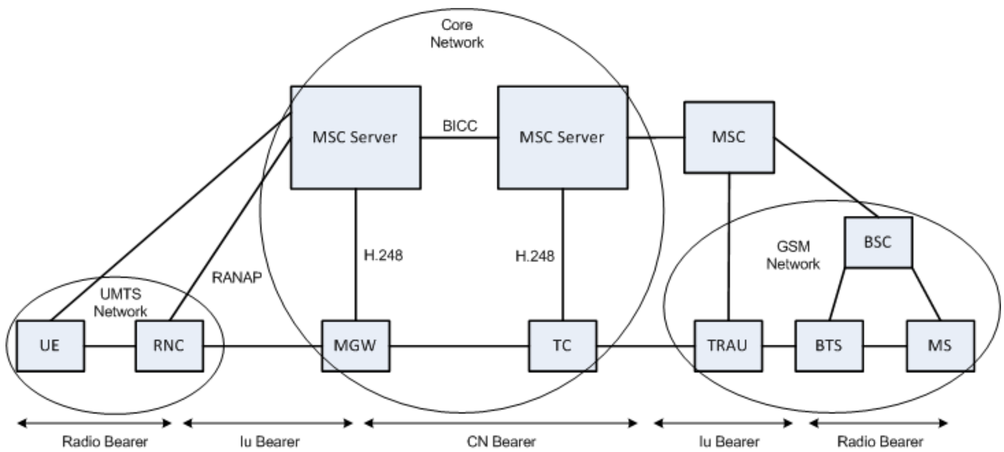

Figure 5.10 : TrFO UMTS-to-GSM architecture

Media Gateway (MGW) Function

In this section the functionalities of Media Gateways will be discussed over the signalling and user data flows between two MGWs in Figure 5.11

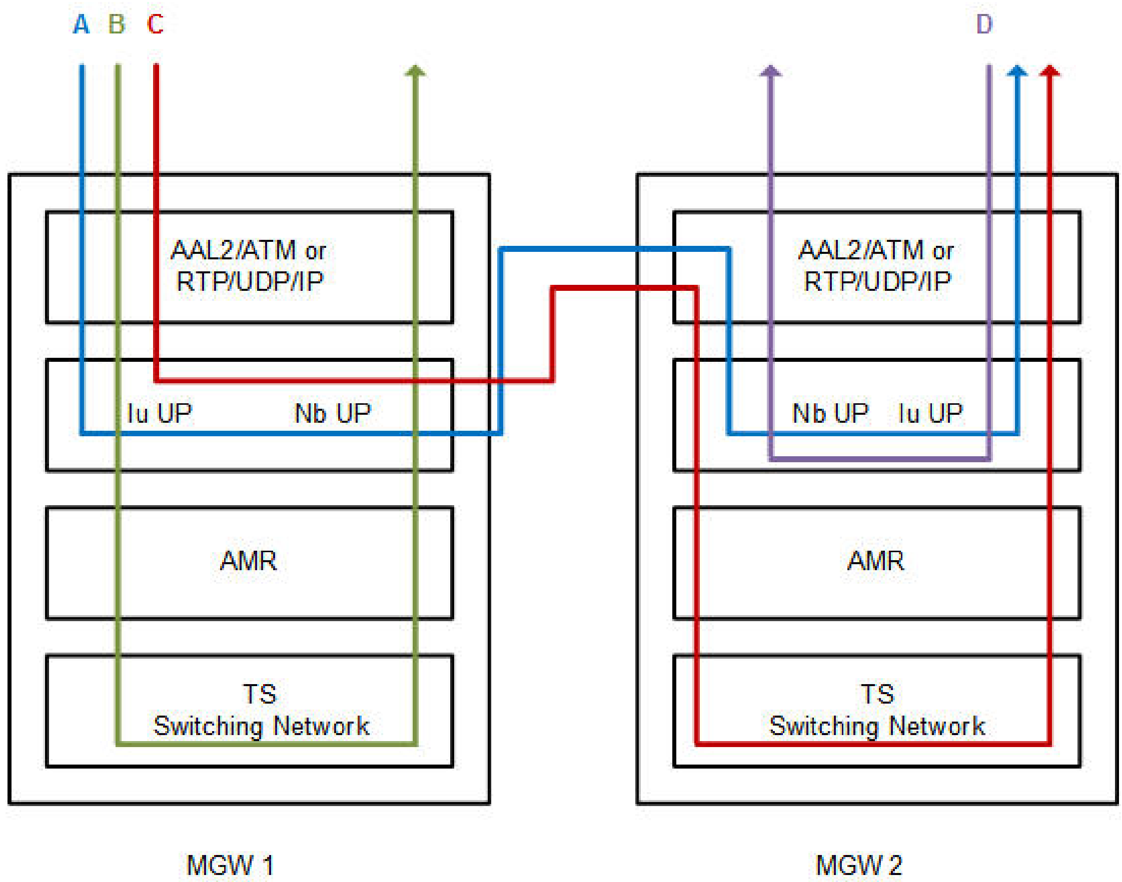

Figure 5.11 : User Plane flow in a TrFO communication

The dark green line (Line B) shows a TrFO call transaction within a MGW. Following this line, it can be seen that user plane medium is accessed via transport mechanism in packet networks either via AAL2 in ATM or RTP and UDP in IP networks. Call flow then follows the uplink procedure which goes through IuUP of the sender and downlink procedure in the other ends IuUP interface and is processed here. As it is also noticed – this process had no business with AMR and/or TDM networks.

Blue line (Line A), corresponds a TrFO user data flow between two MGWs. In this flow it is seen that the user data has been reached over the transport medium and to IuUP interface for the uplink process – is directly sent to NbUP to be transferred to Nb interface.

The data that handled by the first Nb interface is then sent to the other Nb interface. It is assumed in this stage that by the mean time the uplink process is accessed on the other edge of the network too. After the uplink processes have successfully executed in both ends – the downlink period of the sender begins and finally user data reaches the receiver part Iu interface. This whole flow also avoided AMR transcoding and TDM switching.

In addition to the clarified lines in Figure 5.11 in - order to make a comparison – red line (Line C) shows a normal call procedure and purple line (Line D) call flow shows a single MGW flow in TrFO call.

5.3.4 Remote Transcoder Operation (RTO)

Remote Transcoder Operation is a mechanism where just a single transcoder may be needed on the far end of the network. By this perspective RTO is neither a transcoderless operation nor requires tandem transcoding – actually something in between with just a single transcoder in the bearer path. The transcoder arrangement of RTO can be seen in Figure 5.12

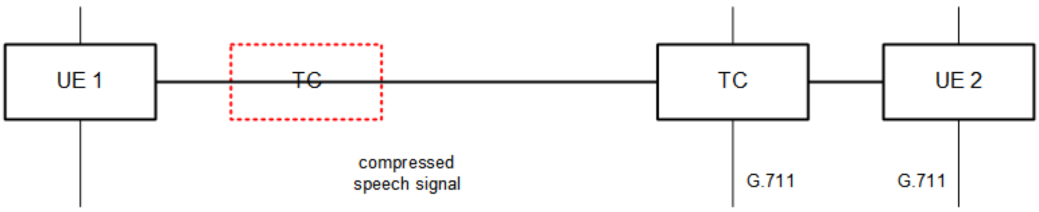

Figure 5.12 : RTO Transcoder arrangement[8]

The RTO shall be enabled in the networks when MS/UEs have incompatible codecs. In the worst case at least one of the transcoding functions would have been bypassed. At the time the transcoder is needed in the network – RTO utilizes a transcoder selected from a prioritized list of transcoders. All the other procedures – except a transcoder in the far end – are more or less identical with Transcoder Free Operations.

On the other hand RTO allows mobile calls from CDMA network to terminate on landline phones. At the far end MGW the samples are extracted from the packet,

[8]To compare with Figures 5.6 and 5.7

transcoded to 64 kbps speech and sent over the circuit switched Inter-Office trunk to the far end office in PSTN [34]

5.3.5 DTMF Handling in TrFO

DTMF communication in TrFO is done always via out-of-band using DTAP procedure. DTMF signals sourced by UE, are sent to O-MSC and then to out-of-band signaling path through the core network. Once DTMF signals are out-of-band signaling path – it is ensured that the signals will be carried to the boundary of the next network.

However in the case that DTMF signals are received from an outside network in-band as I.366.2 profile or RTP payload, Receiving-MGW using H.248 protocol reporting MSC to implement out-of-band signaling transmission through the core network.

5.4 TFO and TrFO Interaction

Today's modern networks may have the capability of accommodating both TFO and TrFO methods aiming a better speech quality, low bandwidth usage and a string of more benefits that in the previous sections have been given in details.

At the very beginning of the call – Network does not know if TFO or TrFO will be possible for the communication. Accordingly transcoder functions are by default allocated. If out-of-band transcoder functionality can succeed to agree on TrFO requirements then transcoders will be disabled and TrFO communication takes place. But at the terms OoBTC functionality cannot agree on the TrFO requirements – transcoders steal some bits from conventional PCM streams and include TFO messages and frames. In the case that distance transcoder can translate the TFO parameters then they start pre-synchronization of IPEs to form a TFO communication. As it can be clearly seen; the decision mechanism belongs to the network and its capabilities in the perspective of using additional specifications.

In case of TFO and TrFO existing in a network together – this is indeed the most modular and practical solution of the recent technologies – allowing them to function in a cycle as the conditions allow. Thinking on a network that has the possibility of enabling both TFO and TrFO disciplines will act according to those major scenarios:

5.4.1 Interaction Scenario A

In this particular example – Although the network supports OoBTC functionality in signaling path however the bearer side in the transit network is TDM and therefore it does not support compressed voice signal. Network must to insert some transcoding functionality.

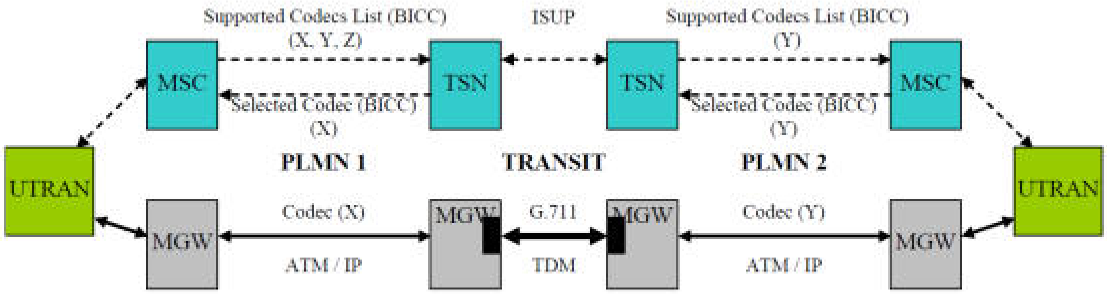

Figure 5.13 : Cascade TrFO & Transcoding Scenario A [37]

It would be better if transit network would have supported compressed speech signal – end to end TrFO could be established. But here the discontinuity in the bearer path will be passed as it is and network will use TrFO on the edge sides. This is really practical. The illustration of the scenario can be seen in Figure 5.13

5.4.2 Interaction Scenario B

Scenario B indicates the situation in case that if only one/PCM codec is available in the end systems.

When OoBTC function can result of the only common codec as PCM - meaning the every network will use any codec that they want and transcoders have to get involved accordingly to make a conversion between them. Basically end-to-end TrFO here is not possible. UMTS-to-GSM internetworking in such condition can be seen in Figure 5.14

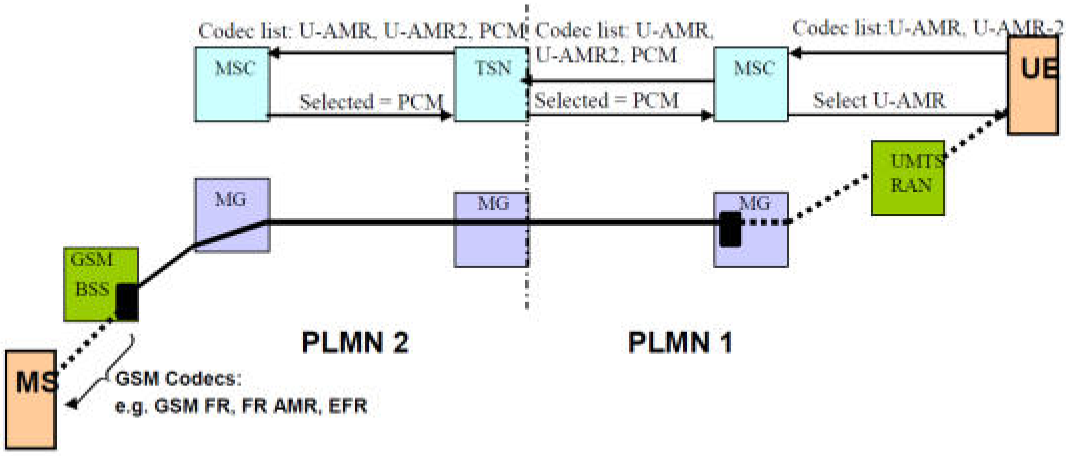

Figure 5.14 : UMTS-to-GSM internetworking in Scenario B [37]

5.4.3 Interaction Scenario C

This is the case which a codec modification on the go is shown in Figure 5.15 – In the normal procedure transcoding functionalities send the Selected_Codec parameter that has been decided OoB, to each other inband. Codec modification can be needed in the case of roaming between the networks and/or can be caused by the selected common codec type with no common ACS.

In this very scenario, codec negotiation is decided, authorized and executed by OoB functionality and then via vertical MGW control protocol H.248, TSN node indicates MGW to initiate and propagate the codec selection

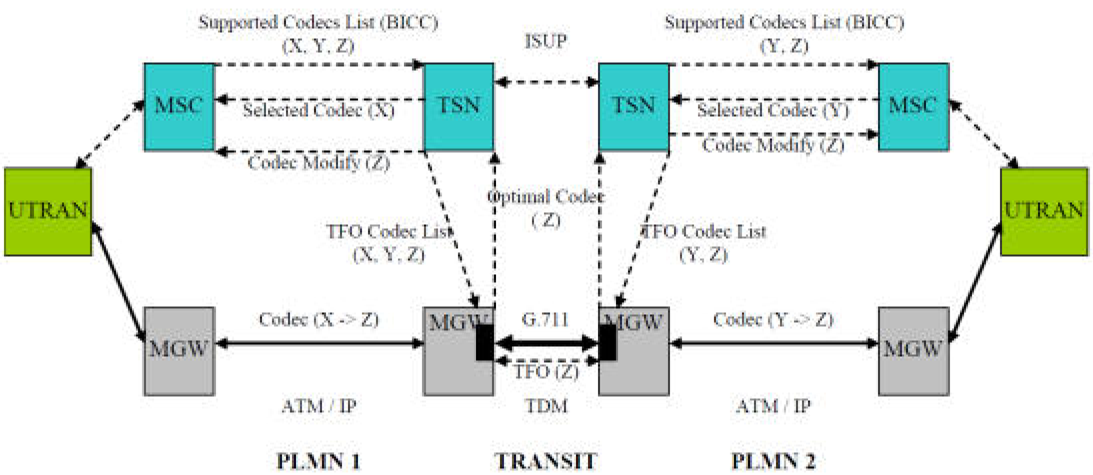

Figure 5.15 : UMTS-to-UMTS call OoBTC supported TFO in Scenario C [37]

5.4.4 Interaction Scenario D

The scenario is a perfect TrFO case as there only one codec used without any intermediate transcoding function. The significance of this scenario is based on codec selection. As it is already studied intermediate network nodes support usually more than one codec and codec selection is defined mainly by end users. The case where AMR-FR codec is common through speech path can be seen in Figure 5.16. Therefore the communication is done TrFO over AMR-FR codec.

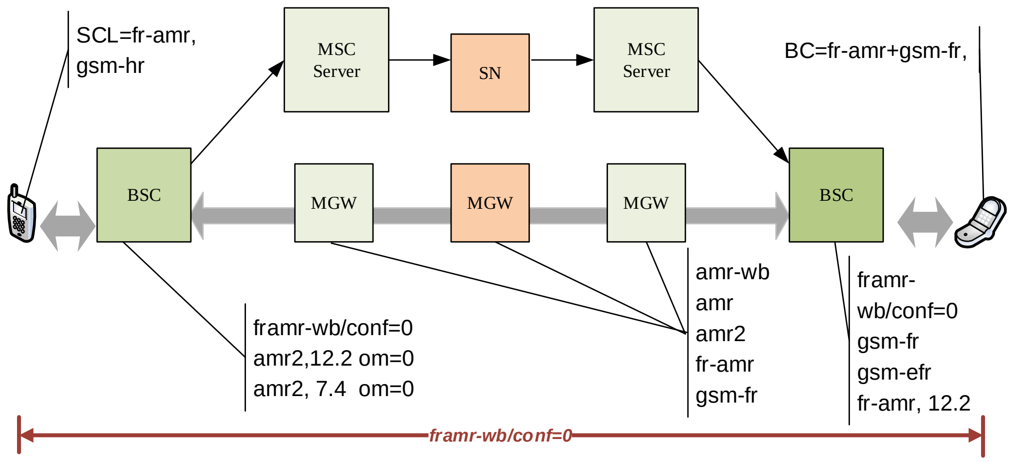

Figure 5.16 : GSM-to-GSM perfect TrFO example [36]

5.4.5 Interaction Scenario E

Scenario E, at the first sight can seem a little bit confusing but the technical reason is fully convincing. IN this scenario although there are two different codecs in the speech path, the communication is done TrFO. The key point here why we don't do transcoding in between two different codecs; is maximum rate control of configuration codes in wiedeband codecs.

Codec Configuration defines a specific set of attributes (done by attendant bits) to a certain codec type. Codec configuration is often defined by *codec/configuration* code as a term. As can be seen in Figure 5.17 two codecs that are used in communication have codec types of '0'. Consequently the codecs configured from 0 to 5 can immediately be fully compatible with each other [35]. In other saying the codecs on the speech path are taken as fully compatible and therefore TrFO can be succeeded.

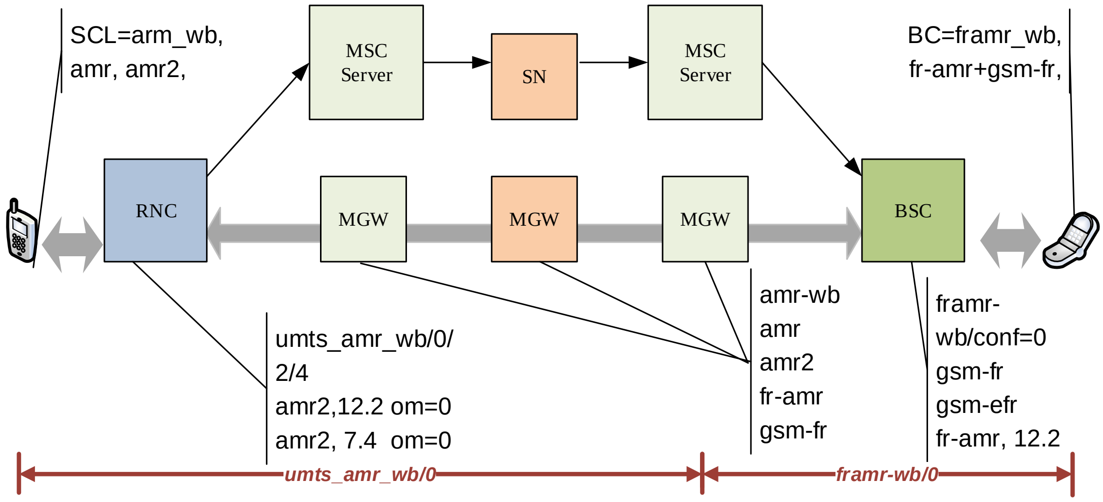

Figure 5.17 : UMTS-to-GSM tricky TrFO example [36]

6 Conclusions

In this work conceptual, architectural and technical aspects of TFO and TrFO have been studied as well as they can be. The Section 5.4 has been created to show if such specifications can really be executed in real life. As it must be also accepted that the laboratory experiments are indeed less complicated and deserted comparing the real life. But nevertheless they are essential aspects of the concept for a deeper understanding of the mechanisms.

My personal aim to do such a master thesis –as explained in the introduction part – to find the answers of the questions come up my mind. I would like to share my answers with the target audience.

Q: Under which circumstances totally bypassing the transcoders are possible?

Total bypass of the transcoders a.k.a TrFO is totally applicable in the real life scenarios. And the circumstances are user plane version (2) of Mobile Networks, common and/or fully compatible codec lists on both UEs and intermediate network nodes.

Q: If so, can transcoders be omitted from the mobile networks for good?

The answer is NO. Unreliable air conditions, RAN interfaces and sometimes mobility itself make the steady TrFO state almost impossible. When the TrFO communication starts; a slight change in the signal degree, network coverage differences etc. result in reversion of TrFO to either TFO or normal means of communication. And in this case usage of transcoders is inevitable.

Q: Would there be the case that we still have to use transcoders?

YES because FOR NOW TFO and TrFO mechanisms cannot be always used. Consequently most of the time normal means of communication must continue.

Q: If transcoder usage is a must then are there any sequential repeating processes that we can get rid of?

The answer is YES. That is why TFO is introduced. In the case where the transcoders cannot be omitted totally, depending on network conditions sequential tandem transcoding processes can be removed from the speech path. That is maybe as not advantageous as TrFO but still gain is a gain.

List of Figures

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List of Abbreviations

3GPP	Third Generation Partnership Project
ACS	Active Codec mode Set
APM	Application Transport Mechanism
BC	Bearer Control
BICC	Bearer Independent Call Control
CC	Call Control
IN	Intelligent Network
ISDN	Integrated Services Digital Network
IuFP	Iu Framing Protocol
MACS	Maximal number of codec modes in the ACS
OM	Optimization Mode
OoBTC	Out-of-Band Transcoder Control
QoS	Quality of Service
RAB	Radio Access Bearer
SCS	Supported Codec mode Set
TFO	Tandem Free Operation
TCH	Traffic Channel
TICC	Transport Independent Call Control
TrFO	Transcoder Free Operation
UP	User Plane

AS	Application Server
BGCF	Breakout Gateway Control Function
BSS	Base Station Subsystem
CDMA	Code Division Multiple Access
CN	Core Network
FDD	Frequency Division Duplex
GGSN	Gateway GPRS Support Node
GPRS	General Packet Radio Service
GSM	Global System for Mobile Communications
HLR	Home Location Register
HSS	Home Subscriber Server
IETF	Internet Engineering Task Force
IMS	IP Multimedia Subsystem
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
ISO	International Organization for Standardization
IWU	Interworking Units
LAN	Local Area Network
MEGACO	Media Gateway Control protocol
MGCF	Media Gateway Control Function
MGW	Media Gateway

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