ECC Report 265

Migration from PSTN/ISDN to IP-based networks and regulatory aspects

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0   EXECUTIVE SUMMARY

The ongoing digital evolution has transformed the world of telecommunications. The demand for new services and increased capacity has rendered existing telecommunications technologies as obsolete in the shift to a ubiquitous IP world. The implementation of IP-based networks, replacing PSTN/ISDN technologies, fundamentally changed many paradigms, notably by allowing the use of a common technology (IP) for the delivery of multiple services (telephony, video, television, Internet, etc.).

The migration in progress towards replacing existing PSTN/ISDN networks with an all-IP platform (both for core and access networks) will modify many parameters, both for operators and end-users. Some changes are already known and solved, while others are still uncertain and could be more difficult to implement. The different stakeholders involved in this migration face many challenges, which basically differ according to their position in the world of telecommunications. One of the fundamental challenges facing regulators is how to enable a fast migration to IP while ensuring a smooth and seamless transition for end-users.

This ECC Report is technical in nature and provides an overview of the current migration to IP-based networks that should enable regulators and other stakeholders to better understand the challenges of this important technological step. It is essentially based on an exchange of experiences in the regulation of migration toward IP-based networks and services between different CEPT member states through their respective national administrations. It also provides a status update on the migration process of some member states and identifies various technical challenges.

The Report does not aim to mandate specific regulatory measures but may be used to inform future policies with the objective of providing a better understanding about the technical and regulatory challenges of migration and some guidance on how this evolution will pave the way for the provision of multiple services on a fast and reliable IP platform.

Chapter 2 gives an account of the different network technologies in chronological order and summarises some of their respective main features in order to better understand the concerns mentioned later.

The main drivers for migration given by the network operators are analysed in detail in Chapter 3 and complex issues behind the migration to all-IP are identified.

Chapter 4 identifies the different migration strategies, either technical or commercial, in correlation to the complexity of the core and the access network, and their interconnection.

In terms of scenario and possible options, different approaches are possible and are identified in Chapter 5. Temporary solution, full migration and a combined approach are all discussed.

Chapter 6 identifies the regulatory aspects to consider when migrating from PSTN/ISDN to new IP-based networks. These regulatory requirements are technology neutral and should therefore be fulfilled when migrating.

The IP-based interconnection for voice services is a complex topic and is detailed in Chapter 7. Types of IP interconnection, signalling protocols, IP codecs, points of interconnection, quality of service, provision of supplementary services, impact of migration on existing telephone numbering ranges and network security are presented.

Chapter 8 examines the different factors affecting the time schedule and the plans for migration to IP-based networks, such as the different migration scenarios, the different challenges of migrating core networks and access networks, current progress of the incumbents and other operators and the number and type of end-users.

Announcements about forthcoming migration projects from PSTN/ISDN to IP-based networks have generated a lot of debate, and indeed emotion, regarding issues like powering, change of paradigm, private networks, non-voice services (alarms, lift alarms, etc.), fax transmission, QoS and DTMF. Chapter 9 considers these different issues and their potential solutions.

Chapter 10 details the conclusions drawn in the report which are:

- One can conclude that, following consideration of the main drivers for migration, the migration from PSTN/ISDN to IP-based networks is unavoidable;

- The obsolescence of the PSTN/ISDN network equipment, the lack of vendor support, the increasing operational and maintenance costs, and the shortage of skilled maintenance staff are the major drivers for migration to IP based networks;

- The migration to IP based networks is necessary to meet the ongoing demand for capacity and multiple communications services;

- Several migration strategies exist but the phased solution is the most often selected because it allows for a softer approach with the existing PSTN/ISDN network working for a period in parallel with the IP-based network components as they are phased in. The analysis presented in this ECC Report suggests that full PSTN/ISDN migration will ultimately prevail over temporary or combined solutions;

- Network operators planning migration projects should take into account any potential impacts on both their wholesale and retail customers and they should clearly communicate their migration plans and allow sufficient time for customers to prepare for the migration;

- In the case of wholesale customers, the migration plans should take account of investments already made and, where necessary, provide alternative equivalent wholesale products;

- Migrating from PSTN/ISDN to all IP may impact the quality of voice services, the performance of networks providing telephony services and the reliability of non-voice services offered via PSTN/ISDN;

- Interconnecting IP-based networks will pose challenges and interoperability issues will arise that may not have been a problem for the interconnection of PSTN/ISDN networks. The diversity of types of IP interconnection, of signalling protocols and codecs requires a harmonised approach, broad consultation among key stakeholders and may require regulatory intervention;

- Depending on national regulatory frameworks, actions from regulators may be required to minimise the risk of any negative impact on customers.

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LIST OF ABBREVIATIONS

Abbreviation	Explanation
2G	Second-Generation wireless telephone technology
3G	Third-Generation wireless telephone technology
3GPP	Third-Generation Partnership Project
3PTY	Three-Party Conference
ACR	Anonymous Call Rejection
ADSL	Asymmetric digital subscriber line
AG / AMGW	Access Gateway / Access Media Gateway
AN	Access Network
ARM-NB	Adaptive Multi-Rate - narrowband
ATM	Asynchronous Transfer Mode
BB	Broadband
BEREC	Body of European Regulators for Electronic Communications
BGC	Border gateway controller
BICC	Bearer-Independent Call Control
BRI	Basic Rate Interface
CAPEX	Capital Expenditure
CC	Country Code
CEPT	European Conference of Postal and Telecommunications Administrations
CD	Call Deflection
CF	Call Forwarding
CH	Call Hold
CLIP	Calling Line Identification Presentation
CLIR	Calling Line Identification Restriction
COLP	Connected Line Identification Presentation
COLR	Connected Line Identification Restriction
CPE	Customer Premises Equipment
CRPD	Convention on the Rights of Persons with Disabilities
CUG	Closed User Group
CW	Call Waiting
DDI	Direct-Dial-In

DNS	Domain Name System
DSL	Digital Subscriber Line
DSLAM	Digital Subscriber Line Access Multiplexer
DSSI	Digital Subscriber Signalling System No. 1
DTMF	Dual-Tone Multi-Frequency
EC	European Commission
ECC	Electronic Communications Committee
ECC WG Nan	ECC Working Group Numbering and Networks
ECM	Error Correction Mode
ECT	Explicit Call Transfer
ETSI	European Telecommunication Standards Institute
ETSI EG	ETSI Guide
ETSI ES	ETSI Standard
EU	European Union
FoIP	Fax over IP
FTTH	Fibre To The Home
GPON	Gigabit Passive Optical Network
GSM-EFR	Basic Rate Interface
HFC	Hybrid Fibre-Coaxial
HGW	Home Gateway
IAD	Integrated Access Device
IC	Interconnection
ICT	Information and Communication Technologies
IETF	Internet Engineering Task Force
IMS	IP Multimedia Sub-System
IP	Internet Protocol
IPSec	Internet Protocol Security
IPvIC	IP-based Interconnection for voice services
ISDN	Integrated Services for Digital Network
ISDN-BA	Integrated Services Digital Network Basic Access
ISDN-PRA	Integrated Services Digital Network Primary Rate Access
ISUP	Integrated Services Digital Network User Part
ITU-T	International Telecommunications Union - Telecommunications
LE	Local Exchange
LTE	Long Term Evolution

MCID	Malicious Call Identification
MDF	Main Distribution Frame
MEGACO	Media Gateway Control Protocol
MGCP	Media Gateway Control Protocol
MOS	Mean Opinion Score
MPLS	Multiprotocol Label Switching
MSAG	Multi-Service Access Gateway
MSAN	Multi-Service Access Node
MSG	Media Service Gateway
NGN	Next Generation Network
NRA	National Regulatory Authority
NSN	National Significant Number
NTP	Network Termination Point
ODF	Optical Distribution Frame
OLT	Optical Line Termination
ONT	Optical Network Terminal
ONU	Optical Network Unit
OPEX	Operating Expenditure
OTT	Over-The-Top
P2P	Peer-to-Peer, Point-to-Point
PBX	Private automatic branch exchange
PCM	Pulse Code Modulation
PESQ	Perceptual Evaluation of Speech Quality
PoI	Point of Interconnection
POLQA	Perceptual Objective Listening Quality Assessment
POTS	Plain Old Telephone Service
PRI	Primary Rate Interface
PSAP	Public-Safety Answering Point
PSTN	Public Switched Telephone Network
PT TRIS	Project Team Technical Regulatory Issues
QoE	Quality of Experience
QoS	Quality of Service
QSIG	Q-Interface Signalling
RFC	IETF Requests For Comments
RIO	Reference Interconnection Offer

RSU	Remote Subscriber Unit
RTP	Real-time Transport Protocol
SAL	Subscriber Access Line
SCCP	Skinny Client Control Protocol (Cisco defined protocol)
SDP	Session Description Protocol
SG	Signalling Gateway
SIP	Session Initiation Protocol
SIP-I	Session Initiation Protocol with encapsulated ISUP
SIP-T	Session Initiation Protocol for Telephones
SS7	Signalling System 7
TCP/IP	Transmission Control Protocol / Internet Protocol
TE	Transit Exchange
TeS	Telephony Server
TDM	Time Division Multiplexing
TDMvIC	Time Division Multiplexing Interconnection for voice services
TISPAN	ETSI's Telecommunication and Internet converged Services and Protocols for Advanced Networking Technical Committee
TMG	Trunking Media Gateway
TSM	Telecoms Single Market
UAM	User Access Module
UN	United Nations
UPS	Uninterruptible Power Supply
USD	Universal Service Directive
USO	Universal Service Obligation
UUS	User to User Signalling
VoBB	Voice over Broadband
VoIP	Voice over IP
VPN	Virtual Private Network
xDSL	Digital Subscriber Line technologies

1 INTRODUCTION

The ongoing digital evolution has transformed the world of telecommunications. The demand for new services and increased capacity has rendered existing telecommunications technologies as obsolete in the shift to a ubiquitous IP world. While classical telephony did not fundamentally change for nearly a century (from late 19th to late 20th century) the first evolution started with the introduction of digital network technologies in the 1980s. The introduction of ISDN networks, still based on a relatively fixed and not very evolutionary infrastructure, was an additional step forwards in the 1990s. The implementation of IP-based networks, replacing the above-mentioned technologies, fundamentally changed many paradigms, notably by allowing the use of a common technology (IP) for the delivery of multiple services (telephony, video, television, Internet, etc.).

The migration in progress towards replacing existing PSTN/ISDN networks with an all-IP platform (both for core and access networks) will modify many parameters, both for operators and end-users. Some changes are already known and solved, while others are still uncertain and could be more difficult to implement. The different stakeholders involved in this migration face many challenges, which basically differ according to their position in the world of telecommunications. One of the fundamental challenges for regulators to solve is: how to enable a fast migration to IP while ensuring a smooth and seamless transition for end-users?

This ECC Report provides a technical overview of the current migration to IP-based networks that should enable regulators and other stakeholders to better understand the challenges of this important technological step. It is essentially based on an exchange of experiences in the regulation of migration toward IP-based networks and services between different CEPT member states through their respective national administrations. It also provides a status update on the migration process of some member states and identifies various technical challenges.

The Report provides a brief historical overview of the various telecommunications technologies and then provides an analysis of the main drivers for migration (mainly due to technological changes and aging/obsolescence). This analysis is based on a recent survey of CEPT member states that has identified different migration strategies and resulting scenarios. ANNEX 2 is published as a separate document to this ECC Report entitled ECC Report 265 – Annex 2.

One cannot discuss the subject of migration to IP-based networks without acknowledging that, from a technical perspective, new methods of interconnecting networks are inevitable. The main characteristics of new interconnection methods are briefly presented for information to better understand some of the challenges inherent in the transition from PSTN/ISDN to IP-based networks.

The migration to IP-based networks is currently underway in almost all CEPT member states but with different approaches and timescales having been adopted. While some countries have already completed the migration process others are still only at the planning phase. Nevertheless, it is anticipated that most CEPT countries will have completed the migration process by the end of the present decade. A summary overview gives an overall picture of the process in different countries.

The document also considers current and future regulatory aspects that need to be considered. This section mainly focuses on end-users, since some features or services (quality of service, powering, location information for emergency calls etc.) could be affected by the transfer of traditional services, such as voice, to IP.

Finally, conclusions are presented which should provide a better understanding about the technical and regulatory challenges of migration and some guidance on how this evolution (if not a revolution) will pave the way for the provision of multiple services, including voice, on a fast and reliable IP platform while retaining some of the essential benefits and characteristics that end-users have come to expect from traditional telecommunications services.

2   HISTORICAL AND TECHNICAL BACKGROUND

The Public Switched Telephone Network (PSTN) has been evolving ever since Alexander Graham Bell made the first voice transmission over wire in 1876. PSTN, is the oldest and most widely used network in the world for the provision of legacy telecommunications services. PSTN was initially designed to provide a short distance connection between any two points or customers[1] (point-to-point). Over time the telephony network evolved to support more customers and endpoints through a network of switches designed to facilitate ubiquitous network connectivity that enabled voice communication over long distances. By placing switching equipment in centralised locations, network engineers were able to interconnect large numbers of end-users via these switches to maximise network access; thus the concept of the circuit-switched network was born.

For many decades, the access network and the core network was analogue, which resulted in long distance calls with poor audio quality (i.e. calls with a low signal level and a high noise level). In order to implement much-needed improvements, network operators started to convert the core network from analogue to digital based on PCM-technology.

In the 1980s, the telecommunications industry began planning for digital services. The industry made network deployment decisions based on the assumption that digital services would follow much the same pattern as voice services, and they conceived a vision of end-to-end circuit-switched services over a network which became known as the Integrated Services Digital Network (ISDN). The focus on ISDN was on transmission of voice signals and low–speed data signals.

With the evolution towards IP-based networks, the circuit-switched network (PSTN/ISDN) began to be migrated towards a new architecture called Next Generation Network (NGN). NGN is based on IP-based platforms (IP multimedia sub-system (IMS), soft switches etc.). The evolution towards NGN is a process in which whole or parts of the existing networks are replaced or upgraded to the corresponding NGN components providing similar or better functionality, while attempting to maintain the services provided by the original network and the possibility of additional capabilities [1].

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[1] In this document the terms subscriber and customer are used interchangeably.

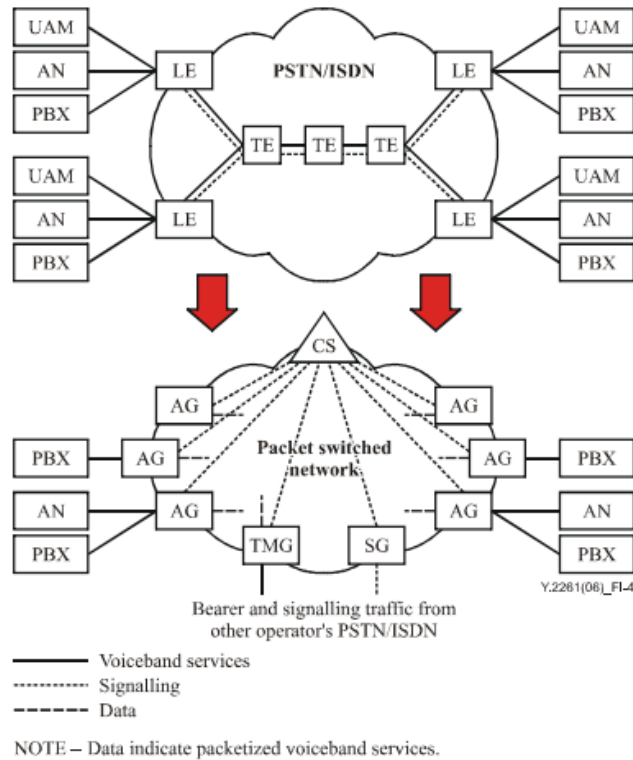**Figure 1: Softswitch-based PSTN/ISDN evolution to NGN [ITU-T Y.2261, PSTN/ISDN evolution to NGN, 2006]**
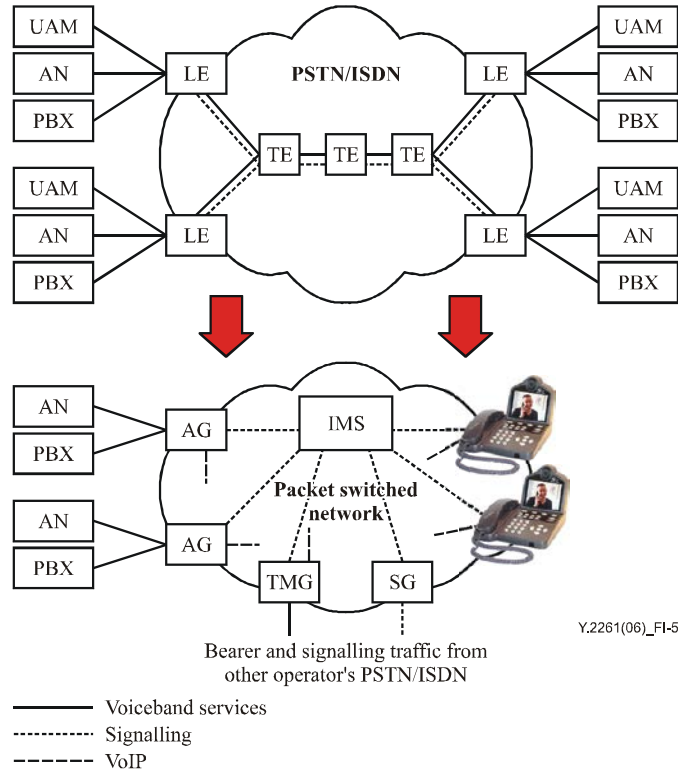

Figure 2: IMS-based PSTN/ISDN evolution to NGN [ITU-T Y.2261, PSTN/ISDN evolution to NGN, 2006]

Figures 1 and 2 illustrate the replacement of PSTN/ISDN solutions by a softswitch-based solution and an IMS-based solution. While a softswitch-based solution involves the replacement of existing TDM switches,

an IMS solution is a multi-layered, all-IP architecture that allows the delivery of voice, data, video and mobile services over a single, packet-switched network.

2.1 PSTN

As defined in ITU-T Rec. G.100 [2], the term "Public Switched Telephone Network (PSTN)" is used for any network providing transmission and switching functions as well as features which are available to the general public, not restricted to a specific user group. The PSTN provides access points to other networks or terminals only within a specific geographical area. From the point of view of an end-to-end connection, a public network can function either as a "Transit Network" (a link between two other networks) or as a combination of "Transit and Terminating Network" in cases where the public network provides connections to terminal equipment such as telephone sets, or PBXs.

PSTN is mainly based on "circuit-switched" technology which establishes a dedicated communications channel (by circuit) between two nodes through the network before the nodes may communicate. The circuit provides the fixed bandwidth according to the channel size (bandwidth) and remains connected for the duration of the communication session. The PSTN network consists of the transmission, switching, signalling and intelligent networks. The transmission network enables the transmission of all kinds of traffic (voice, video and data). It consists of nodes called multiplexers and links among multiplexers. The switching network enables switching of traffic between the sender and the appropriate destination. A switching network consists of switches. The term "switch" describes the ability to cross-connect a phone line with many other phone lines and switching from one connection to another. A switching network operates in a connection-oriented mode. That means that prior to enabling the exchange of traffic between users, there is a need to reserve resources on the path between the sender/origination point and the receiver/termination point. To reserve resources, all switches on the path exchange signalling information. In the case of circuit-switched technology, the Signalling System 7 (SS7) protocol is most commonly used which implements out-of-band, common channel signalling where signalling information is carried in a dedicated channel which is separate and distinct from the bearer channels. Intelligent network functionality is used in the voice network for the provisioning of services such as freephone, premium rate, virtual private network, account card calling, etc. It consists of a set of application servers containing service logic and service data.

The main features of the PSTN may be summarised as follows:

- Mainly voice-band services (voice and 3.1 kHz audio-band data services);
- The channel remains reserved and cannot be used by competing users (even if no actual communication is taking place);
- Provides continuous transfer without the overhead;
- A dedicated path persisting between two communicating parties or nodes can be extended to signal content; and
- A constant delay during a connection.

To provide telephone services to all users nationally and/or internationally, the PSTN is organised in a hierarchical structure. There are several different models, but the basic idea is similar such as providing connectivity to end-user (by Local Exchange), providing connectivity between nodes at the city level (by Tandem Switch), providing connectivity between different regions (by Toll Switch) and, finally between countries (by International Gateways). Each level of the hierarchy has a different role for providing connectivity. Relevant systems such as switching and transmission systems are equipped with different technologies and capabilities.

2.2 ISDN

Integrated Services Digital Networks (ISDN) is a network that provides digital connections between user-network interfaces identified with a set of communication standards for simultaneous digital transmission of voice, video, data, and other network services over the traditional circuits of the PSTN. ISDN is a 64 kb/s channel (called B channel) based on the circuit-switched telephone network system (offering circuit-switched connections for either voice or data) while providing access to packet networks for data.

The key feature of ISDN is that it integrates speech and data on the same lines, adding features that were not available in the PSTN. ISDN channels are delivered to the user in one of two pre-defined configurations:

▪ Basic Rate Interface (BRI) - provides 2 B-channels (bearer channels) at 64 kbit/s each and 1 D-channel (delta channel) at 16 kbit/s. The B-channels are used for voice or user data, and the D-channel is used for any combination of data, control/signalling, and X.25 packet networking. The 2 B-channels can be aggregated by channel bonding which provides a total data rate of 128 kbit/s (in both upstream and downstream directions). The BRI ISDN service is commonly installed for residential or small business services;
▪ Primary Rate Interface (PRI) - provides a varying number of channels depending on the standards in the country of implementation. In Europe it is 30 x B-channels + 1 x D-channel on an E1 2.048 Mbit/s according to ITU-T Recommendation G.704 [3]. One timeslot on the E1 is used for synchronization purposes and is not considered to be a B or D channel.

Unlike the BRI and PRI ISDN, which are often called narrowband ISDN (N-ISDN), ITU also defines a broadband ISDN (B-ISDN). B-ISDN is based on a switching technique known as Asynchronous Transfer Mode (ATM). The fast-packet technology of ATM makes it possible to transport information with widely differing characteristics, from bursty data services to high-definition television signals, over a range of speeds. It manages the establishment of point-to-point and point-to-multipoint connections through the switched network. It also supports on-demand, reserved, and permanent services, as well as connection-oriented and connectionless services. The B-ISDN supports much higher data rates (above 2Mbit/s) and has a packet switching orientation.

The benefits of ISDN may be summarised as follows:

▪ Uses the existing telephone wiring system (twisted pair-lines) without incurring additional costs and enhances wide area networks usage (voice and non-voice applications);
▪ Provides integrated access to telephone services, circuit-switched data and digital video by using the telephone network;
▪ Significantly faster call setup compared with PSTN-based telephony;
▪ Provides a faster data transfer rate than can be achieved using modems by using the B-channel including where multiple B-channels are bonded; and
▪ Supports simultaneous calls on one access line.

2.3 IP

Voice over IP (VoIP)[2] is the real-time transmission of voice signals using the Internet Protocol (IP) over the public Internet or a private data network. Legacy PSTN circuit-switched voice networks are evolving to packet-switched networks, where VoIP can be offered by two different approaches:

▪ Next Generation Network (NGN): ITU-T Y.2001 [4] defines the next generation network (NGN) as follows: NGN is a packet-based network able to provide telecommunication services and able to make use of multiple broadband, QoS-enabled transport technologies and in which service related functions are independent from underlying transport related technologies. NGN transports voice over managed and secured IP networks with well-defined guarantees for customer reachability, communication quality, reliability and connectivity, while supporting services inherited from the PSTN.
▪ Over the Top (OTT) Voice services (formally called Voice over Internet): Internet telephony with voice traffic routed on a "best effort" basis as a customer or peer-to-peer application on top of the Internet (unmanaged networks), providing only limited service quality.

From a telephone service provider's perspective, a "best efforts" approach to service quality for end-to-end IP-based infrastructure is not appropriate as Internet calls share bandwidth on the network with other competing applications. Therefore the NGN approach is a more suitable and appropriate way of routing and delivering voice traffic in publically available electronic communications networks.

The steps involved in originating VoIP telephone calls are similar to traditional telephony and involve signalling, channel setup, digitalisation of the analogue voice signals, and encoding. Instead of being

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[2] VoIP is used in many applications (even within gaming). This report focuses on telephony services only.

transmitted over a circuit-switched network, however, the digital information is packetised, and transmission occurs as IP packets over a packet-switched network. VoIP systems employ session control and signalling protocols to control the signalling, set-up, and tear-down of calls.

Communication on the IP network is perceived as less reliable in contrast to the circuit-switched PSTN because it does not provide a network-based mechanism to ensure that data packets are not lost, and are delivered in sequential order. The QoS issue is covered further in this document.

3 MAIN DRIVERS FOR MIGRATION

The PSTN was originally designed to offer fixed telephony services only. The increasing demand for Internet access and the requirement for higher bandwidth to serve all the IP-based services, coupled to the trend in most countries showing a significant shift from fixed to mobile telephony and the inability of existing networks to support all these services simultaneously, leads to the need for higher capacity and more powerful IP-based networks.

Existing PSTN/ISDN voice networks are slowly reaching end of life. The exhaustion of stocks of spare parts, cessation of support for network equipment and software by the major vendors and human resource issues caused by the retirement of engineers with the skills and experience to support legacy systems have all added impetus for a faster migration process. The cost of maintaining each PSTN/ISDN customer is inevitably rising. Simultaneously, operator revenue from these services is declining as they become of less importance to end-users who have a wider choice of communications applications to choose from including e-mail, IM, video chat and social media. Therefore, from the operator's perspective, the motivation for migration is to lower costs and exploit revenue generating opportunities. NGN equipment is cheaper to purchase and to operate a common platform for multiple service offerings which are based on IP. While the operator's core concerns are to reduce costs, increase revenue and remain competitive they must also take account of their regulatory obligations in their respective migration strategies such as maintaining specific (high) QoE/QoS that users have come to expect from the PSTN/ISDN.

The following sections analyse in more detail the main drivers for migration.

3.1 MARKET TRENDS

With fewer customers for fixed telephony, it makes sense for operators to migrate to a more cost-efficient service platform. The NGN architecture with centralised telephony servers provides lower operational costs per customer compared to the legacy PSTN/ISDN. In addition, by moving to IP-based networks it is possible to integrate telephony with other services which represent revenue generating opportunities (e.g. Internet access and TV services) by essentially offering a larger service portfolio to customers via the same fixed access line.

3.2 LEGACY EQUIPMENT

Legacy equipment is becoming obsolete and is becoming less suitable for providing services in the new IP networks as the demand for higher speed and capacity increases. Equipment suppliers, who are also keen to exploit new revenue growing opportunities, are gradually withdrawing support for legacy equipment and software. There is a possibility that the operator can sell the legacy equipment to developing countries still using PSTN/ISDN networks and another option is the recycling of existing equipment.

3.3 VENDOR SUPPORT

In some cases, the planned lifetime of the equipment has already been exceeded. Most equipment vendors' PSTN portfolios, maintenance and system support contracts with operators are slowly expiring or will reach end-of-life by 2020. Operators or service providers who are not already making plans and being proactive about migration will struggle with upgrades, maintenance and lack of spare parts for PSTN/ISDN. They may also have to pay a premium to equipment vendors for extended equipment and software support for legacy products. Given the increase in the number of operators who are already implementing migration strategies, the need for PSTN/ISDN equipment spare parts is further reduced which further increases the risk of higher premiums for extended support for those who are late in implementing their migration strategies. As the uptake of new NGN equipment gathers pace, economies of scale are being exploited as the price of NGN equipment decreases and the decision to proceed with migration sooner rather than later becomes an easier one for operators.

3.4 OPERATIONAL COSTS

The process of transformation to IP-based networks may not involve the decommissioning of existing copper infrastructure although it will usually involve extending the network of optical fibres at least to the cabinet level. Some operators will opt for full IP transformation, which will include the use of copper access loops for analogue signal for speech. Traffic may then be switched to an IP-based soft-switched core at the exchange-based MSAN. The decommissioning of the copper network may be part of a plan that has nothing to do with an upgrade to an optical fibre infrastructure or migration to IP-based network because many operators want to break down unprofitable parts of their copper networks for reasons of cost reduction. This is especially the case in rural areas. As already mentioned earlier, operating expenses (OPEX) for PSTN/ISDN network are increasing and service providers are supporting a declining number of customers. However, the PSTN/ISDN equipment remains, as do the high costs required to power, maintain, and house it. The combination of low port utilisation and high OPEX is creating unsustainable per-user costs and the need to run parallel networks for voice and data services only exacerbates the problem.

3.4.1 Maintenance costs

IP-based networks are considered to be simpler and cheaper to maintain than PSTN/ISDN. There is a reduction in the number of nodes to manage and it is possible to facilitate integration across fixed and mobile voice platforms. Systematic migration reduces the cost and risk of maintaining aging equipment. By unifying and consolidating network infrastructures, operators can phase out certain parts of the network. They can also reduce the number of spare parts and vendors they have to manage. Migration to NGN reduces the amount of equipment and the requirements for power and cooling of the more up-to-date NGN technology (chipsets and software) are lower.

3.4.2 Real estate costs

Accommodating PSTN/ISDN equipment represents a significant cost. The equipment takes up considerable floor space and is typically distributed over many locations. A potential challenge for PSTN/ISDN operators is to reduce the real cost of housing the equipment while, at the same time, maintaining the number of PSTN/ISDN customers it has and the revenue it generates from them. Migration can help in addressing this challenge in addition to boosting capacity and supporting new subscription offerings. IP-based network nodes require 2-5 times less space when compared with existing PSTN/ISDN equipment. IP-based networks also support efficient indoor configurations and service centralisation, both of which can further help to save space. If operators release large amounts of space, they may consider selling or renting that same space to create an additional revenue stream. Therefore it is a realistic assumption that the real estate costs for IP-based networks will be significantly lower than those for PSTN/ISDN.

3.5 STAFF COMPETENCE AND HUMAN RESOURCES

There are two basic points to consider regarding staff competence and human resources:

- A large PSTN/ISDN network with network equipment installed over a wide geographic area requires a certain number of specialised engineers to maintain that equipment. Moreover, the equipment may come from a number of different vendors which inevitably requires those engineers to have the skills to maintain different equipment. The complex maintenance of PSTN/ISDN as outlined in section 3.3.1 therefore requires a higher human resource commitment when compared to IP-based network equipment;
- As the production of PSTN/ISDN equipment has decreased the availability of specialised employees, and the availability of training on PSTN/ISDN equipment have correspondingly decreased. This has led to a reduction in the number of skilled employees available that can respond to market demands.

Therefore, the human resource costs related to the maintenance of IP-based networks is also expected to be lower than for PSTN/ISDN as there will be fewer network nodes and fewer sites to maintain. Furthermore, as IP technology is standardised globally, the required engineer skillset is applicable to equipment provided by different vendors.

4 MIGRATION STRATEGIES

When it comes to the migration strategies from PSTN/ISDN to all-IP, there are two general approaches.

In general, migration of the core network is less challenging than migration of the access network, because there is significantly less service disruption to the customer. Therefore most operators introduce all IP into the core network first, and then expand to the access network, with the option to have a period of parallel running of both the PSTN and all-IP core networks. There are various ways of migration from legacy networks to all-IP and the choice depends on per country characteristics and the individual operator's situation. In general, the approach will be whether to overlay or replace the existing network or a combination of both.

Another aspect of the migration process relates to how PSTN/ISDN networks are interconnected and the implications this will have for interconnecting IP-based networks. The question of the number of interconnection points is important and will remain an open issue from a technical, economical and regulatory point of view. Supposing that the number of interconnection points is reduced and limited to the IP core network locations if required, it can be assumed that the termination at a PSTN user cannot always be realised locally. The configuration of the services and the planned centralisation of control functions will have an impact on the question of which locations the traffic from other networks can be taken over from or handed over to and how alternative service providers can integrate their services into an NGN platform.

The most common approach to migration towards IP-based networks is a phased solution which means that the existing PSTN network will work for a while in parallel with the IP-based network components as they are phased in. It is a reasonable solution for many reasons. First of all, CAPEX for the immediate replacement of the PSTN represents a significant financial outlay and that is the case even with a step-by-step migration, where the initial outlay required for the core IP-based network is still very high. The investment required is proportionate to the size of the service provider and the geographic scale of the network. Legacy networks in such cases will eventually be replaced, but it can take several years or decades. The geographical scale, the number of different technologies used and prevailing market conditions will require operators to pass through many small steps until they reach their desired objective, which is a fully migrated IP-based network.

An alternative to a phased (step-by-step) migration approach is to swap out all of the existing network components from PSTN/ISDN technology to IP technology in a very short time. This approach is only feasible for competitors of the incumbent operators who have a relatively small core network and only a minimal access network.

4.1 TECHNICAL MIGRATION

PSTN/ISDN is a legacy platform that has prevailed for decades and in which a big investment has been made. The services provided on this network remains a major, albeit declining, source of revenue for operators. Therefore it is logical that operators wish to maximise their revenue stream from existing investments while also focusing on evolving and further developing the architecture for voice services to exploit other revenue-generating opportunities.

The existing PSTN/ISDN architecture is complex and consists of different networks and network layers which have been designed mainly for the provision of voice services and which operate in parallel to other networks and network layers for the provision of data services. The maintenance of these parallel networks results in higher operating costs. The migration process normally starts with changing the technology of the core network from PSTN/ISDN to all-IP where the link to the access network is emulated using equipment (access media gateway) that enables the translation between the IP and the PSTN/ISDN protocols (using media gateway controller functionality). Some operators may choose to maintain both the PSTN/ISDN and all-IP core networks running in parallel such that the PSTN/ISDN core network controls traffic of customers served through the legacy access network, while the all-IP core network controls the traffic of customers served through MSANs and/or full broadband connections. .

4.1.1 Technical solution

The technical solution decided upon for the migration process will depend on a number of variables that require careful consideration including the scale of the existing network infrastructure, market trends, expectations of management, the strategic plans of the service providers, commercial realities that need to be addressed such as the proliferation of Over The Top services (OTT) and, ultimately, the cost of deployment.

In order to decide upon the most optimal migration approach, the following solutions are possible (see Chapter 5 for more details):

- A phased solution in which elements of the existing network architecture are replaced with IP-based technology on a phased basis in accordance with a migration plan;
- Full PSTN migration to IP-based network where all users are migrated;
- Combined approach of a phased solution and full migration for different parts of the network.

4.2 COMMERCIAL MIGRATION

Migrating from PSTN/ISDN to IP-based network infrastructure is first and foremost a commercial business decision for operators. One can assume that the decision to migrate will ultimately be a profitable one for incumbent operators but the timing of the migration will have an impact on the rollout cost. Incumbent operators must decide if it is financially viable, in terms of the level of operational expenditure, to keep providing services to customers using the existing network or to make the necessary capital investment in a complete transformation of the network.

Migration is a necessity for operators in order to succeed in today's competitive market as they must be responsive to trends in the market place in order to exploit new revenue-generating opportunities. Choosing the right time in which to start the migration process is a critical decision because at some point providing services on legacy networks will become significantly more expensive when compared to the provision of services via an all-IP infrastructure. One can assume with a high degree of certainty that the transition to all-IP will result in OPEX savings (e.g. cost of power consumption, staff and equipment maintenance). These savings will vary from operator to operator depending on network scale and customer numbers. Therefore the optimal start time of the migration process is critical as the risk of incurring losses increases if the migration process is delayed or too premature in the case of an overwhelmingly high percentage of subscribers using only voice services.

Operators need to assess whether the legacy network is removed immediately or whether a layered approach or step-by-step migration is more appropriate. For services provided through the existing network, operators need to consider if the cost of providing these services would be higher after the migration to an IP-based network infrastructure.

Customer retention is a key consideration in the migration decision. The cost of migrating each customer may vary in different parts of the network depending on the number of customers and the profit margin for each subscription. In some remote and less developed parts of the territory, where there may be a smaller number of users, and especially where those users only subscribe to a voice service, the operator will eventually have to decide whether to enter into the migration process or to discontinue the provision of services to these customers. If the operator does decide to terminate services to these customers they should inform their subscribers accordingly and provide information and guidance on how customers can switch to alternative service providers while allowing a sufficient period of time for the switching process to take place. When the operator is the designated undertaking to provide access at a fixed location as a universal service, it remains responsible towards these customers through the universal service obligation.

5   SCENARIO ANALYSIS AND POSSIBLE OPTIONS

Through scenario analysis, it is possible to evaluate different migration options and interim solutions. In order to determine the optimal way to do this in order to achieve the desired architecture with regard to the technical and financial impacts, the following are possible options:

- Do-nothing scenario in which no action is taken and no investment is made. This approach can lead to deterioration in the quality of services provided and is likely to result in a loss of customers and, consequentially, a loss of revenue. This scenario is not considered as a realistic option;
- Temporary solution, in which changes to the existing architecture or replacement of individual parts of the network such as transit switches (but not local switches) take place rather than a full migration to a single network and IMS platform;
- Combined approach, where different parts of the network are migrated, on a full or temporary basis, depending on the requirements and characteristics and potential revenue generating opportunities in those areas;
- Full PSTN/ISDN migration to NGN IP architecture[3]. A complete replacement of the PSTN/ISDN with a common platform (IMS) where only one remaining network exists and to which all PSTN/ISDN users are migrated. The access options to cater for different types of customers are being covered in Section 5.2 below.

5.1   TEMPORARY SOLUTION

In this solution, only parts of the PSTN/ISDN are replaced by new technology while other parts are maintained without changes. A typical solution involves the replacement of legacy transit exchanges with soft switches and media gateways which are then connected via an ATM-based or IP-based network. All local exchanges and concentrators (remote units) are maintained with existing circuit-switched TDM technology. This solution is illustrated in the Figure 3 below.

There are several benefits with this solution, including the need for less equipment, less power consumption and less space requirements for transit nodes. Also, a large number of legacy TDM transmission links in the transit network are replaced by high capacity ATM-based or IP-based connections. This solution can also be implemented over a relatively short period of time since it does not involve changes in the access network and there is no service disruption for customers.
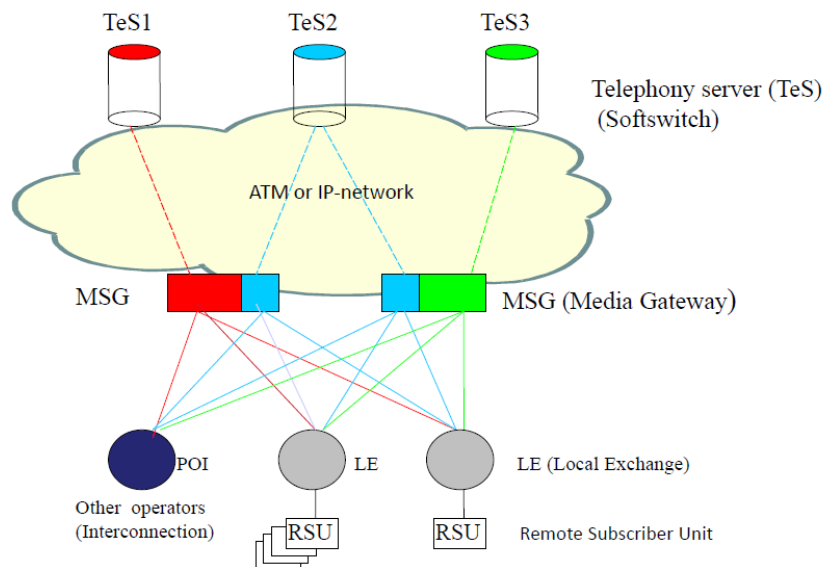

Figure 3: Temporary solution, example from Norway based on Ericsson ENGINE platform

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[3] NGN IP architecture is the common term used in technical standards and specifications.

5.2   THE COMBINED APPROACH

In some scenarios, a combination of approaches can be used to achieve migration. The migration strategy can be implemented following an area-by-area analysis and defining in which areas customers can be migrated to all-IP and which areas customers can continue to be served by legacy platforms.

For the latter, the connection of customers to the all-IP core network can be established in two ways:

- Via old PSTN switches, which most likely need to be upgraded; or
- Via soft switches.

This solution is very complex from an operational and maintenance point of view as different types of equipment must be implemented and maintained in the network. The installation and maintenance of this equipment requires a broad skillset. The drawbacks of this approach include:

- Customers cannot always be provided with the same set of services; and
- It is expected that the costs will be higher mainly due to costs for maintenance contracts and increased human resource costs for operation and maintenance.

Based on these drawbacks, it is clear that the combined approach requires careful evaluation.

5.3   FULL PSTN/ISDN MIGRATION

The ultimate goal of the PSTN/ISDN migration to all-IP is to use the same access network to provide different services. The objective is to have a network architecture that is end-to-end IP-based. This enables the reuse of common control and service layers for different types of access (copper, FTTH, 2G/3G/LTE, etc.), limiting costly adjustments for those layers. Provisioning of voice via broadband line for 2-Play and 3-Play customers presents significant savings, since one port is used for all services. However, for voice-only PSTN/ISDN customers, a number of options are possible.

5.3.1   Emulation of PSTN service (migration to MSAN POTS card)

A multi-service access node (MSAN), also known as a multi-service access gateway (MSAG), is a device typically installed in a telephone exchange (although sometimes in a roadside serving area interface cabinet) which connects customers' telephone lines to the core network, to provide telephone, ISDN, and broadband (such as DSL) all from a single platform [5].

Services are provided for broadband customers by aggregating multiple channels of information, including voice and data, across a single shared access link using an integrated access device (IAD). MSAN POTS ports are used for voice-only customers. Migration using MSAN POTS is illustrated in Figure 4 below. In case of a power failure the user device will have power supply through the MSAN POTS card.

The provisioning of voice services via MSAN POTS for voice-only customers allows for a faster and smoother migration. Customers are remotely provisioned with no work required at the customer premises. Equipment costs are also lower as (remote provisioning—no in-house work) no IAD is necessary for voice-only customers. However, no new services can be provided to the customer since the MSAN POTS card is only capable of emulating the PSTN voice service. The MSAN includes functionalities which are described in TISPAN as access media gateway.
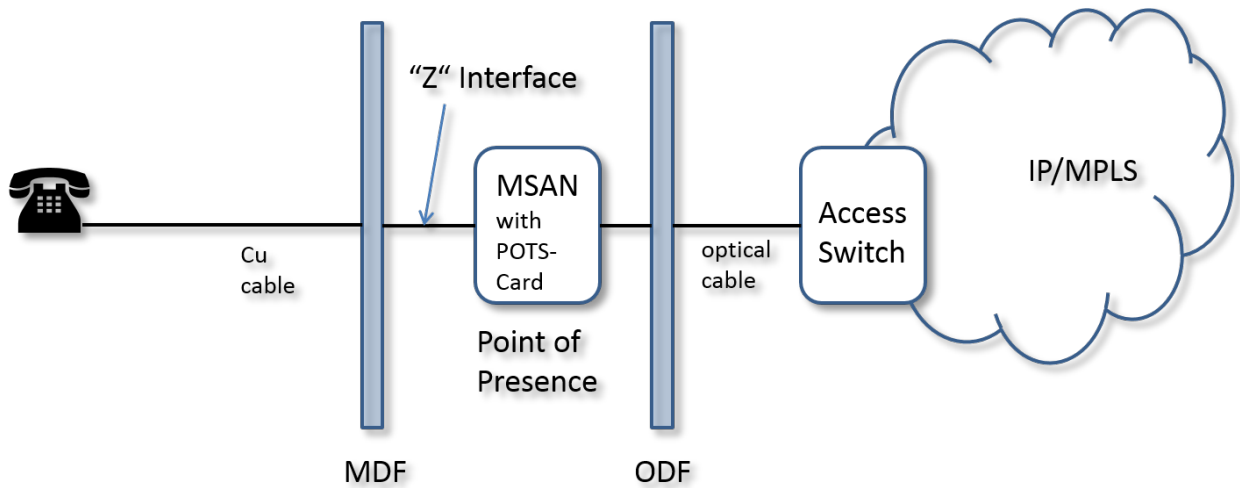
Figure 4: Migration to MSAN POTS card

5.3.2 Migration to broadband port

This option represents full PSTN migration connecting all users via broadband ports to the IMS. As high and sufficient broadband penetration is required, PSTN migration should be planned as an integral component within the broadband access transformation strategy. The MSAN concept is one solution to follow this plan. The capital investment for this approach is obviously higher when compared to MSAN POTS as it is likely that there will be a need for work to be carried out at the customer premises. If the existing installed broadband capacity is not high enough then additional investment is needed for broadband ports.

There are two main technical solutions. The first solution is suitable in situations where the access terminates at the customer premises using copper and the second where the access terminates at the customer premises using optical fibre. These two solutions are detailed further below. For certain niche markets, some operators may also consider the use of wireless access.

a) VoBB over metallic access (copper access)

In this solution, the xDSL modem or Home Gateway (HGW) is installed at the customer's premises, and connected via MDF to an xDSL port of the MSAN. The MSAN is then connected via an optical cable to the operator access point which is at the edge of the IP/MPLS network carrying the traffic for all services. This is illustrated in Figure 5 below.
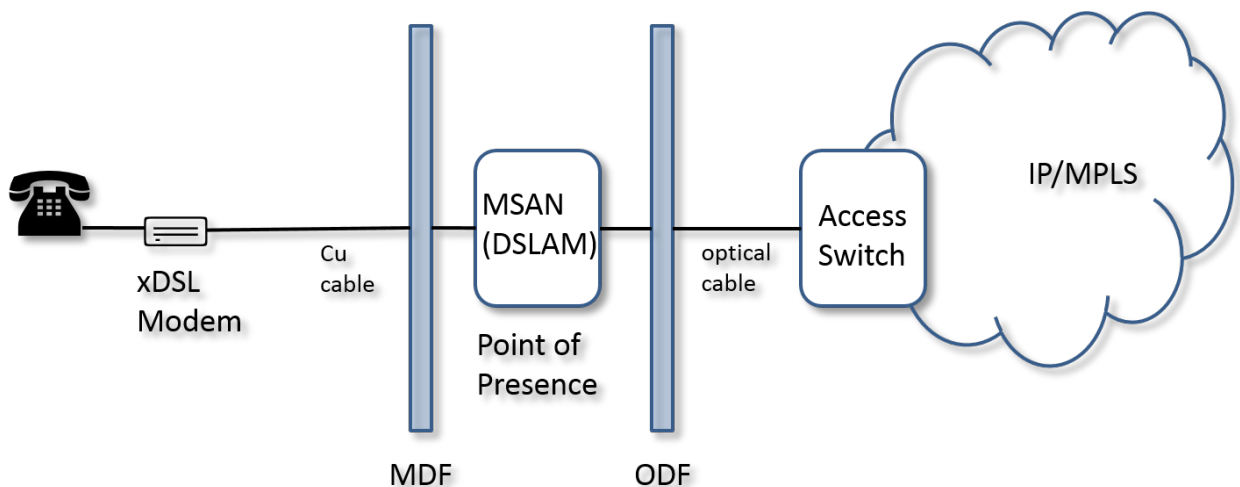

Figure 5: Migration to broadband port – Cu access

In order to migrate to full IP, operators must build networks with optical cables. In parts of the network where they have not yet rolled out optical fibre access, the MSAN POTS card and VoBB solutions (depending on the service required by the users) may be used. Therefore, the above-mentioned options are considered as temporary solutions and in the longer term a full migration to all-IP will be required.

b) VoBB over optical access

Different optical access technological solutions are used by operators, some are point-to-point and others are point-to-multipoint based. The main advantage of point-to-multipoint solutions like GPON is the potential savings that can be made in building cable infrastructure as GPON requires fewer optical fibres to be installed. With this approach the Optical Line Termination (OLT) is installed on the operator side while the Optical Network Unit (ONU) or Optical Network Terminal (ONT) is installed at the customer's premises. This is illustrated in Figure 6 below.
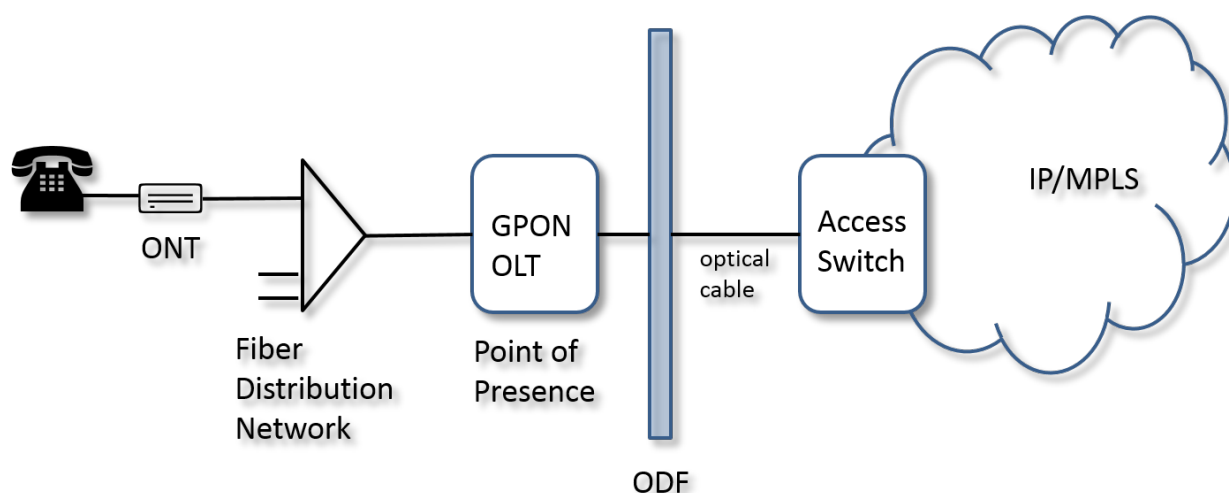
Figure 6: Migration to broadband port - GPON

5.3.3 Access of ISDN BA/PRA customers

Migration for traditional ISDN-BA services requires IADs which are installed at the customer premises and connected between the modem and the end-user ISDN device. IADs interwork with an IMS platform. This solution typically requires a technical intervention at the customer's premises. In some cases, ISDN-BA subscriptions can also be migrated without a technical intervention at the customer's premises in the case where the customer has only a voice service and wants to keep and ISDN phone. In these cases the voice service use switches to the IMS platform through MSAN ISDN-BA ports.

ISDN-PRAs that were not replaced timely with another commercial offering will be migrated without needing a technical intervention at the customer's premises using a MGW-solution (Media Gateway) which interworks with an IMS platform. Figure 7 below describes the various migration scenarios for ISDN services.
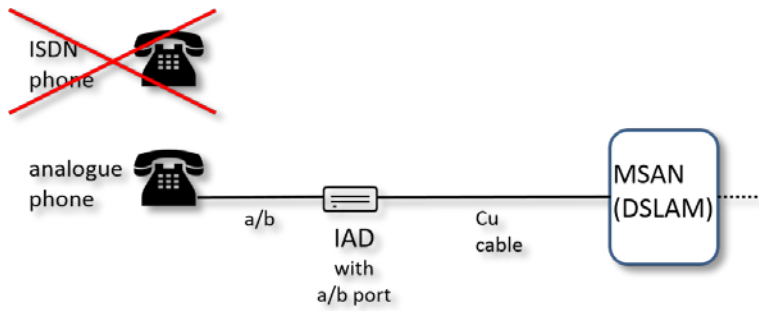
ISDN BA

Voice over BB – Migration to ordinary IMS product (one line)
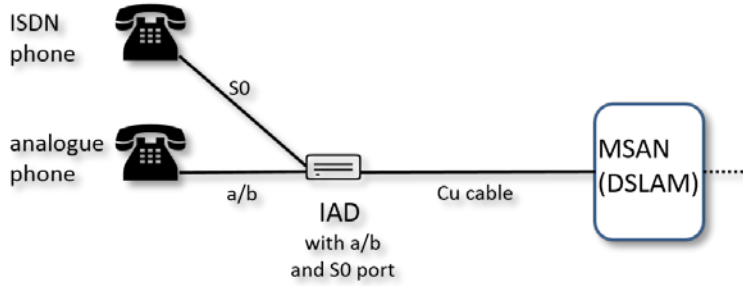
ISDN BA

Voice over BB – ISDN BA VoIP gateway with one SO and up to 2 FXS ports
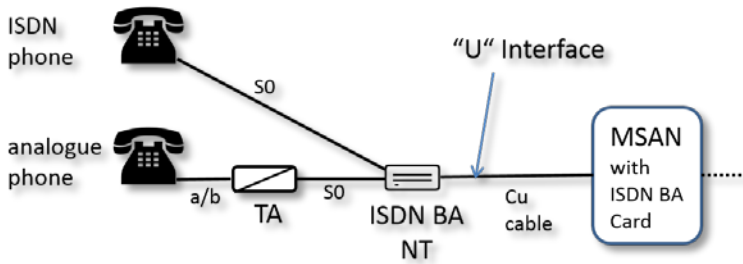
ISDN BA

ISDN BA service (MSAN line card)
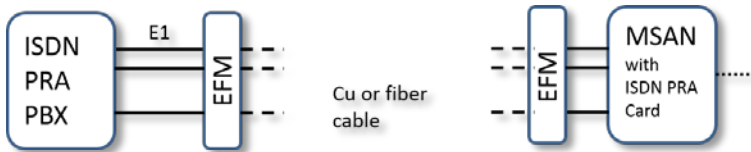
ISDN PRA

ISDN PRA PBX connected with multiple E1 links

Figure 7: Migration of ISDN BA/PRA customers access

6 REGULATORY ASPECTS

PSTN/ISDN and publicly available telephone services are subject to regulatory requirements set out in national and European legislation[4]. These obligations include *inter-alia* universal service obligations, interconnection obligations, obligations to provide access to emergency services and to provide caller location information for emergency calls, number portability obligations, lawful interception obligations as well as obligations to ensure the security and integrity of networks and services. In general, these regulatory requirements are technology neutral and should therefore also be fulfilled when migrating to new IP-based networks.

6.1 UNIVERSAL SERVICE OBLIGATION (USO)

6.1.1 Legal background

For EU Member States, regulatory requirements for USO are laid down in Directive 2002/22/EC (*"on universal service and users' rights relating to electronic communications networks and services"*) as amended by Directive 2009/136/EC - "the Universal Service Directive (USD)" [6].

Article 4.1 of the USD states that *"Member States shall ensure that all reasonable requests for connection at a fixed location to a public communications network are met by at least one undertaking".* Article 4.2 states that "*the connection provided shall be capable of supporting voice, facsimile and data communications at data rates that are sufficient* [emphasis added] *to permit functional Internet access, taking into account prevailing technologies used by the majority of subscribers and technological feasibility."*

Minimum requirements for data rates "sufficient" for Internet access are not specified further.

Article 2c of the USD defines a "Publicly Available Telephone Service" as a *"service made available to the public for originating and receiving, directly or indirectly, national or national and international calls through a number or numbers in a national or international telephone numbering plan."*

Consequently, voice services that are not available through national/international numbering plans are not considered as publicly available telephone services.

The USD also includes requirements concerning quality of service aspects (QoS). Article 22 of the USD requires Member States to "*ensure that national regulatory authorities are […] able to require undertakings […] to publish comparable, adequate and up-to-date information for end-users on the quality of their services […]*". Article 22 also states that "*regulatory authorities may specify […] the quality of service parameters to be measured*" and that "*authorities are able to set minimum quality of service requirements on an undertaking or undertakings providing public communications networks*"

6.1.2 Voice over broadband

Historically, USO requirements have been focused on telephony with Internet access as an "add-on" feature. However, with migration from PSTN/ISDN to IP-based networks, telephony services can be implemented and offered to the end-user in different ways (see Chapter 5).

For customers who are connected via fibre, xDSL or modern cable-TV (HFC) networks, there are normally no capacity limitations for using VoIP services.

6.1.3 Future USO requirements

Several countries have already introduced USO for broadband, e.g. Belgium, Croatia, Finland, and Switzerland. But at the same time, these countries have maintained USO for publicly available telephone services even if most available broadband services are capable of carrying voice with good quality. However, not all broadband technologies are suitable for voice communications. Broadband via satellite introduces a

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[4] In September 2016 the European Commission published proposals for amending the regulatory framework for electronic communications in the EU. At time of writing, these proposals are under discussion and therefore not addressed in this report.

significant time delay which may degrade service quality. In general, broadband connections with low capacity (below 1 Mbit/s) may give an unreliable service quality for voice.

But as the capacity for commercially available broadband services continues to increase, the USO minimum requirement for broadband is also likely to increase in the years to come. Finland increased its USO obligation from 1 Mbit/s to 2 Mbit/s in 2015 and other countries are also considering an increase of the minimum capacity requirement. With guaranteed capacities at 2 Mbit/s or higher, practically all types of VoIP services can be used with good quality.

Future USO requirements are therefore likely to be more focused on broadband, rather than telephony, since VoIP services can be carried over (all) broadband connections that fall within the USO definition. If basic broadband and/or mobile coverage is guaranteed through new USO requirements it can be foreseen that the present USO requirements for telephony only will gradually be relaxed or perhaps entirely abandoned.

6.2 NETWORK AND SERVICE PROVISION RELATED REQUIREMENTS

6.2.1 Freedom of choice for users

Regarding freedom of choice for users, the relevant EU legislation imposes obligations on access to, and interconnection of, electronic communications networks and associated facilities, as well as on facilitating a change of provider.

Article 12 of the Directive 2002/19/EC ("Access Directive") as amended by Directive 2009/140/EC [7] states that:

Operators may be required:

> *(e)   to grant open access to technical interfaces, protocols or other key technologies that are indispensable for the interoperability of services or virtual network services;*

> *(g)   to provide specified services needed to ensure interoperability of end-to-end services to users, including facilities for intelligent network services or roaming on mobile networks;*

> *(h)   to provide access to operational support systems or similar software systems necessary to ensure fair competition in the provision of services;*

> *(i)   to interconnect networks or network facilities.*

This article is amended by Directive 2009/140/EC in the sense that:

> *"(a)   …access to specified network elements and/or facilities, including access to network elements which are not active and/or unbundled access to the local loop, to, inter alia, allow carrier selection and/or pre-selection and /or subscriber line resale offer*

> *(j)   to provide access to associated services such an identity, location and presence service"*

The facility to changing service provider is guaranteed by Directive 2009/136/EC [8] amending Article 30 of the USD and stating that *"subscribers with numbers from the national telephone numbering plan who so request can retain their number(s) independently of the undertaking providing the service*".

These various obligations have been developed and implemented in a PSTN/ISDN context (circa 2000-2010) and were generally well respected. Since 2010 the migration to IP-based networks has been taking place throughout Europe and it is not expected that this will lead to the need for substantial amendments to the existing legislation as these obligations can easily be respected when the PSTN/ISDN is replaced by IP-based technology.

However it is necessary to monitor various points as the ETSI document entitled "*Regulatory Requirements and Public Interests in NGN standardisation (SR 080 005)*" [9] recommends:

▪ Special attention should be drawn to the regulatory aspects of mobility in its various forms (nomadicity, handover, fixed mobile convergence) including the management of mobility;

- It has to be ensured that necessary user information is transferred in a NGN network in order to provide other operators and third parties with an accurate identity and location information for their services and applications;
- The numbering topics such as usage of E.164 numbers, number portability, and carrier pre-selection in NGN should be continued;
- It should be clarified how the NGN standardisation could contribute to the EC Recommendation on Next Generation Access e.g. open access interface on layer 2.

6.2.2 Quality of Service requirements

Taking in account that Quality of Service (QoS) regulation has to be technologically neutral there are not any specific requirements and parameters measurements which are different from traditional (PSTN/ISDN) networks. The general approach to QoS is specified by the USD in Articles 21 and 22. In practice, the common measurement parameters for telephony and end-to-end connectivity can be set by national legislation as set out in Article 22 of the USD. This subject is discussed in detail in ECC Report 195 [10]. The most important parameters for voice telephony are: unsuccessful call ratio, call set up time and speech transmission quality described by ETSI EG 202 057-1 [11], ETSI EG 202 057-2 [12], ITU-T Recommendation P.862 [13] and ITU-T Recommendation P.863 [14]. From the customer point of view, these parameters are appropriate for the voice telephony service independent of the technology used to provide this service.

Monitoring QoS end-to-end control on regular basis can provide evidence regarding the consistency of the QoS and allows for a comparison of the QoS levels before, during and after migration from PSTN/ISDN to IP-based networks. Potential challenges regarding QoS degradation are described in Section 9.6.

Another QoS parameters measurement methodology may be used and typical parameters for Internet access services may be measured. These parameters are: transmission speed, delay, delay variation, packet loss ratio and packet error ratio as described by ECC Recommendation (15)03 [15] and ECC Report 195, ETSI EG 202 057-4 [16] and ITU-T Recommendation Y.1540 [17]. Such kinds of measurements may be useful in cases where the QoS degradation is identified while analysing voice telephony end-to-end QoS measurement results to find the potential causes of QoS level degradation.

6.3 SECURITY AND PRIVACY-RELATED REQUIREMENTS

The analog and digital telephony (PSTN/ISDN) world was principally designed as a closed entity, without interaction with external networks. As a result the security of equipment and the personal data was virtually guaranteed and reinforced by the national character of the networks. IP-based networks represent rather an opening to the global world and multiply the interactions between national and transnational networks and services, thus increasing the risks in terms of security and privacy protection. The legislation currently in place for these aspects may require adjustments to correspond to the new context of the all-IP telecommunications world.

6.3.1 Security

In a world increasingly interconnected one essential component is the security of the networks and services. This is a complex and very current subject because of the parallel development of criminal activities directly affecting the security of IT and telecommunications infrastructure. ICT security is the subject of many ongoing research activities as key stakeholders in government and within the industry seek to put the necessary protections in place to guarantee a very high security for essential services in a world of increasingly interconnected networks.

From an EU perspective, the subject of security has been addressed in the relevant legislation frameworks and is becoming more central in terms of future policy considerations.

EU Directive 2002/21/EC [18] ("Framework Directive"), as amended by Directive 2009/140/EC, addresses the issue of security. In particular, Article 13a of amending Directive 2009/140/EC [19] addresses the essential concerns of *security and integrity*. It states that:

"Member States shall ensure that undertakings providing public communications networks or publicly available electronic communications services take appropriate technical and organisational measures to

appropriately manage the risks posed to security of networks and services. Having regard to the state of the art, these measures shall ensure a level of security appropriate to the risk presented. In particular, measures shall be taken to prevent and minimise the impact of security incidents on users and interconnected networks".

Furthermore, it states that *"Member States shall ensure that undertakings providing public communications networks take all appropriate steps to guarantee the integrity of their networks, and thus ensure the continuity of supply of services provided over those networks".*

For access to numbers and services, Article 28 of Directive 2009/136/EC [20] amending the USD and Privacy Directive 2002/58/EC [21] contains requirements that ensure that the relevant authorities are able to require undertakings providing public communications networks and/or publicly available electronic communications services to block, on a case-by-case basis, access to numbers or services where this is justified by reasons of fraud or misuse.

6.3.2   Privacy protection

Various essential aspects are to be considered in terms of privacy protection (Directive 2009/136/EC amending Directive 2002/58/EC).

First of all the provider should guarantee the security of processing by:

- ensuring that personal data can be accessed only by authorised personnel for legally authorised purposes;
- protecting personal data stored or transmitted against accidental or unlawful destruction, accidental loss or alteration, and unauthorised or unlawful storage, processing, access or disclosure; and
- ensuring the implementation of a security policy with respect to the processing of personal data.

The provider must take appropriate technical and organisational measures to safeguard security of its services, if necessary in conjunction with the provider of the public communications network with respect to network security. In case of a particular risk of a breach of the security of the network, the provider must inform the subscribers concerning such risk (Article 4 - Directive 2009/136/EC).

The confidentiality of the communications and the related traffic data must be ensured. In particular, listening, tapping, storage or other kinds of interception or surveillance of communications and the related traffic data by persons other than users is prohibited, without the consent of the users concerned, except when legally authorized to do so. Storing information, or gaining access to information already stored, in the terminal equipment of a subscriber or user is only allowed on condition that the subscriber or user concerned has given his or her consent (Article 5 - Directive 2009/136/EC).

Traffic data relating to subscribers and users processed and stored by the provider must be erased or made anonymous when it is no longer needed for the purpose of the transmission of a communication. For the purpose of marketing electronic communications services or for the provision of value added services, the provider may process the data to the extent and for the duration necessary for such services or marketing, if the subscriber or user to whom the data relate has given his or her prior consent. Users or subscribers shall be given the possibility to withdraw their consent for the processing of traffic data at any time (Article 6 - Directive 2009/136/EC).

Where presentation and restriction of calling and connected line identification is offered, the service provider must offer the calling/called user the possibility, using a simple means and free of charge, of preventing the presentation of the calling/connected line identification on a per-call basis (Article 8 - Directive 2002/58/EC).

Where location data other than traffic data can be processed, such data may only be processed when they are made anonymous, or with the consent of the users or subscribers to the extent and for the duration necessary for the provision of a value added service. Users or subscribers shall be given the possibility to withdraw their consent for the processing of location data other than traffic data at any time (Article 9 - Directive 2002/58/EC).

Subscribers are given the opportunity to determine whether their personal data are included in a public directory (directories of subscribers), and if so, which, to the extent that such data are relevant for the

purpose of the directory as determined by the provider of the directory, and to verify, correct or withdraw such data (Article 12 - Directive 2002/58/EC).

Concerning unsolicited communications, the use of automated calling and communication systems without human intervention (automatic calling machines), facsimile machines (fax) or electronic mail for the purposes of direct marketing may be allowed only in respect of subscribers or users who have given their prior consent (Article 13 - Directive 2009/136/EC).

6.3.3   Lawful Interception

The obligation for lawful interception is considered to be technology neutral. Therefore operators will need to retain their capability to intercept traffic following the migration to IP-based networks.

6.4   CUSTOMER IMPLICATIONS DURING MIGRATION

Customer protection is a sensitive issue particularly relevant in relation to the migration to all-IP. All necessary information must be provided to users in order to ensure a smooth migration and total respect for their elementary rights, e.g. pricing, influence of power supply, access to the emergency services with accurate location information and routing of calls or guaranteed and easy access to eAccessibility services.

6.4.1   Customer protection

During the process of PSTN/ISDN migration to all-IP, certain problematic issues related to service features might arise. Generally, customers are unaware of underlying technological changes to the network but when failures occur a lack of communication only serves to intensify the resistance of customers to change. Therefore, it is necessary to have a comprehensive communications strategy and provide clear and concise information to affected customers before and during the migration project. This applies for both residential and business customers using the most appropriate and effective channels of communication. It should be clearly explained to customers what CPE is needed during and after migration to IMS. Unless operators are contractually committed that current tariffs and other features that the customer uses should remain the same, in the event that there are any changes to tariffs or service offerings, these should be clearly communicated in advance in accordance with any applicable legislation.

In cases where CPE equipment needs to be replaced, the impact of this change on user experience needs to be communicated to customers and they should be adequately informed about any other potential impacts of the migration. Any change of service features should be described in terms & conditions and service specifications. From the customer's perspective the migration should be easy and seamless and if there are any envisaged service disruptions then they need to be communicated well in advance of the start of the migration project. The migration process should preferably not result in the customer incurring any expenses whether financial or otherwise. Regulators have a role to play in this regard.

For business customers using ISDN services connected to a PBX, the operator must take care to provide such customers with necessary information about actions that should be carried out in order to use the service after migration. This information should be provided to business customers well in advance of the start of the migration project to allow business customers to make preparations and contingency plans.

PSTN services typically continue to work during power outages as the CPE is powered from the local exchange. However, modern alternatives usually require backup power to keep operating. It is known that customers are mostly not familiar that when using IP-based voice, electricity is required to power the user terminal. Operators should inform customers that IP-based voice services will be unavailable in the case of a power outage at the customer's location and that the only possible solution to use at least voice services is to have backup power and an Uninterruptible Power Supply (UPS - for more details see Chapter 9.1). This information is particularly important for customers because they must know that in such situation emergency services will not be available through the fixed network.

During the migration period, customers will expect assurances that their services won't be interrupted and they should get real-time support to help resolve any issues that are likely to arise. Also, a special focus during the migration should be placed on wholesale customers because of the complex processes of negotiation and the complexity of the products.

6.4.2 Emergency communications

In many countries IP-based telephony services provided to the public have to fulfil the same obligations as PSTN or ISDN-based telephony. Concerning the required emergency communication, the ECC Report 193 [22] outlines the regulatory situation in various CEPT countries.

Session Initiation Protocol (SIP) is a communications protocol commonly utilised for VoIP. SIP has the advantage that with its Geolocation Header it can submit more location information to the PSAP during call setup than with PSTN/ISDN. But like many other IP-based communication protocols SIP has, in case of nomadic use, a problem with the location estimation specifically when location based routing of emergency calls to the right PSAP is required. It could even happen that a PSAP with regional responsibility receives an emergency voice call coming from a location outside the PSAP's country. This problem with nomadic use is addressed in more detail in Chapter 8 of the ECC Report 225.

Two possible technical solutions are:

1. To rely on location information estimated and provided by the UE. This is the approach IETF is following. The main issues with this approach are that not all UEs support it and that the location information can be easily manipulated by the user;
2. That the access provider (lower layer access network and Internet access provider) and voice service provider cooperate to estimate the location of the caller in case of an emergency call. A standardised solution is under development. ETSI (specifically the expert group which works on the EC standardisation mandate M/493) worked out a functional architecture for such a solution and approved it as a European Standard in February 2015. Work on the protocols for the individual interfaces of that architecture is still in progress and expected to be finalised in 2017.

ECC Report 225 concluded that *"it should be possible to significantly improve on the provision of caller location information for nomadic VoIP services once the M493 standardisation work has been successfully completed within ETSI"* but it also concludes that *"even with the ratification of appropriate standards, location of emergency callers using VoIP technology may not be as reliable and effective as for traditional fixed-line services today"*.

6.4.3 eAccessibility

eAccessibility refers to the ease of use of information and communication technologies (ICTs), such as the Internet and telephony services, by people with disabilities. For a blind person the most convenient way of receiving a message is commonly an audio signal (or a vibrating keyboard or matrix). For a deaf person it might be a video signal or a text message. Other sectors outside of electronic communications, e.g. public ticket machines and many others may also benefit from these techniques.

The technical evolution in voice and video recognition and synthetisation, the growing speed of multimedia processors and the automation of translation services are central contributors for increasingly accessible communications systems in the future. Human intervention is still the base for the majority of accessibility services. The ongoing ICT evolution will further enhance access to communications services for an increasing number of people with disabilities. This is a central concern for a society as it seeks to be more inclusive when it comes to marginalised sectors of the population, including people with disabilities and the elderly.

The most important types of support offered by ICT products and services are in this context:

- Mobility, for movement-impaired persons, using mobile devices offering the possibility of remotely contacting other persons and services or remotely controlling some devices;
- Translation of voice messages (for deaf or hearing-impaired persons) to:
  - Text, like in voice telephony relay services or subtitling on TV programming, and
  - Sign language, commonly used in video relay and TV programming.
- Translation of text to audio signals, for blind or visually-impaired persons.

In general, the worldwide legislation and regulation derives from UN convention on the Rights of Persons with Disabilities (CRPD) [23] also applicable in European countries.

There is a European Union policy to support the development of accessible tools and services. The Universal Services Directive, as amended by Directive 2009/136/EC, requires public communications (e.g. TV, telephony) to be accessible. The implementation of these accessibility requirements differs from country to country. But associating accessibility requirements with USO creates an obligation on operators with universal service obligations to provide eAccessibility services such as:

- Relay services converting text interface messages to voice and vice-versa (for voice telephony);
- Subtitling services offering on a display a text reproducing the audio messages (in the case of TV or video messages in general, expectable in the future also for video telephony);
- Sign Language offered on a display by an overlapped image of the (human or 'avatar') partner reproducing the audio messages (in the case of TV or video messages in general);
- Audio description of a video environment to allow a blind receiver to perceive the context of the transmitted video message.

The migration from PSTN/ISDN to all-IP helps to support eAccessibility because of the higher flexibility and increased number of communication means of IP-based networks.

7   IP-BASED INTERCONNECTION FOR VOICE SERVICES

Interconnection is the physical and logical linking of two or more communication networks to ensure end-to-end service connectivity and to enable customers of interconnected operators to establish communications with each other. Interconnection between PSTN networks is well established and there are little or no interoperability issues. This is because PSTN interconnection makes use of the same signalling system (usually SS7), the same audio codec (usually G.711), the same numbering scheme (E.164), the same media transport (TDM) and the same interfaces (E1/T1)). As operators in EU Member States are in the process of migration from PSTN/ISDN to IP-based networks there are good technical reasons to implement IP-based Interconnection for voice services (IPvIC) as it will eliminate the need for conversion of traffic from IP to TDM and all voice traffic can stay completely on IP.

The BEREC report "Case Studies on IP-based Interconnection for Voice Services in the European Union" [24] shows that the type of operator which most often offers IPvIC is the other fixed network operators followed by the fixed network incumbents and the mobile network operators. Analysis also shows that, from an overall perspective, the IPvIC conditions (based on the experiences of ten EU Member States) are very similar but characteristics may differ reflecting some specific national circumstances.

The regulatory and major technical characteristics of IPvIC are described in detail in the following sections.

7.1   REGULATION REGARDING IPVIC

Some European countries imposed regulatory obligations to offer IPvIC on fixed and mobile operators on the termination markets. Usually, large operators are obliged to publish a Reference Interconnection Offer (RIO). This obligation may not be imposed on smaller operators if it is found to be too heavy a burden. Several countries included this obligation on the origination markets also, while transit is included in the RIO on a voluntary basis as it is no longer regulated.

Relevant operators and stakeholders are involved in the process of defining the national specifications as it is important that operators agree upon how the voice interconnection based on IP will be done. Technical characteristics of IPvIC can be defined in decisions, recommendations or in technical specifications issued by the relevant NRA or ministry, depending on a country's specificities, but there are also examples where they are agreed by an industry forum/body.

Regulatory intervention regarding the transitional period for the migration to IPvIC may be needed as operators may have different opinions on when migration from TDMvIC to IPvIC should be done. It depends on how far they have come with the migration of their networks to an NGN or IP-based network. If they already finished the migration process, they may want to migrate the voice interconnection to IP within the shortest possible time as existence of two parallel interconnection solutions equates to higher costs. Operators that have only started migration of their networks to NGN or IP-based network, or have not started yet, may want to prolong the phasing out of TDMvIC. Regulation could provide balance to ensure a smooth transition in these cases.

Also, there may be a need to regulate the minimum period of notice regarding the phasing out of TDMvIC to allow operators to prepare themselves. Besides the minimum period of notice for technical shutdown of TDMvIC, a minimum period of notice could be also defined for commercial closure of TDMvIC so that operators are informed when old interconnections or capacity extensions will no longer be available. Also, if there are changes in interconnection fees for the remaining operators this should be communicated in advance.

An additional aspect of the regulation could be to delay the migration to IPvIC following a comparison with the migration plan as this can have negative impact on operators. Until now, only one country has defined an economic disincentive for delaying the migration process.

7.2 MAJOR TECHNICAL CHARACTERISTICS OF IP-BASED VOICE INTERCONNECTION

The major technical characteristics are already stated in the BEREC report, but the report does not provide a full technical description and comparison of IPvIC characteristics and parameters. The BEREC report treated themes such as "general characteristics of the IPvIC" and "technical characteristics of the IPvIC" and analysed EU markets and market participants in defined countries. This analysis categorized market players into incumbents, other fixed networks and mobile network operators.

To transmit voice signals between different operators' networks it is significant for the technical parameters of interconnection to be compatible for these operators' networks. This section covers interconnection aspects, as it is illustrated in Figure 8 below (middle part). This section gives additional information about major IPvIC technical characteristics, based on the BEREC report data, and provides an analysis of technical problems and standards. It does not provide statistical information, which is already summarised in the BEREC report.

The migration phase from PSTN to IP-based networks is still ongoing in many countries. While some operators' networks are already fully based on IP, or are in process of full network transition to IP-based networks, other operators still use PSTN for voice and fax services. Where different technologies are used, connecting two separate networks becomes a problem. This problem is solvable but results in additional expenses for network operators using the PSTN network to transmit voice.

There are a variety of possible IP-based networks technical solutions for interconnection (IC). However, some challenges can arise, for example, on issues such as interconnection point placing, use of different signalling protocols, applicable IP audio codecs and supplementary service provisioning. Resolving these problems are no less important in order to avoid incompatibility issues between different operators' networks which can lead to a degradation in quality of service. In many countries, especially during the migration period from PSTN to IP-based networks, telephony service providers are obliged to ensure IP telephony quality parameters correspond with PSTN telephony quality parameters on the same level or better. Therefore, the same basic services should be provided to the end-user at the same level of quality or better.
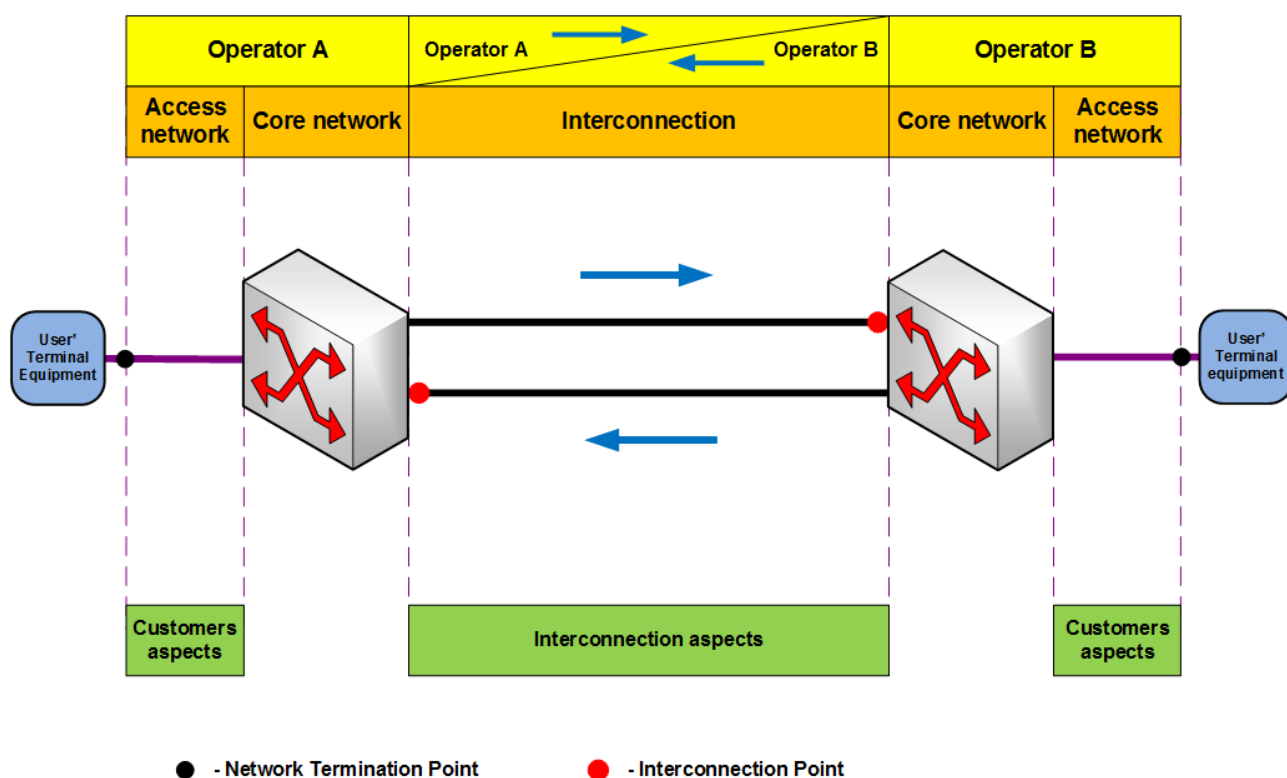

Figure 8: General Network structure

In Figure 8 above, the Blue arrows illustrate the call direction. In most cases the interconnection point is defined in call termination point. In the case of a typical network structure with two interconnection points, interconnection can be one-sided or two-sided. The figures in this section show only the two sided situation.

7.2.1 Types of IP-Based Voice interconnection

There are two basic IP-based voice interconnection types that could be used: direct interconnection using physical links and direct interconnection using the public Internet, i.e. virtual interconnection. Both of these types are in use today. There is a high probability that virtual interconnection will be used more often in the not too distant future.

Currently, in some EU Member States, small operators already connect their networks using virtual interconnection. Incumbent operators mainly use direct physical interconnection links. The reason for this is that incumbent network operators and other big network operators already have a fully functioning physical network infrastructure and the migration from PSTN to IP-based networks can be implemented using this existing infrastructure. New smaller operators do not have this type of infrastructure deployed, and because of that interconnection using public Internet is a better and more affordable option.

As transit services are also based on interconnections, this subject is also discussed in this section.

One of the questions that arises in IP-based networks interconnection is the separation of voice traffic from overall data traffic. Operators may provide not only voice services but also Internet access services. However, call origination and termination fees are defined for voice services only and so it is essential that voice traffic is separated from data traffic before the interconnection point.

7.2.1.1 Direct interconnection link

Point-to-point (also known as P2P) retains the same structure as PSTN interconnection via a direct physical interconnection link. This direct interconnection link is illustrated in Figure 9 below which shows two switches with a direct physical link between them. The schematic structure of interconnection stays basically the same for PSTN and IP-based networks interconnection with the only difference being that PSTN interconnection typically uses SS7 signalling whereas IP-based networks interconnection typically uses SIP signalling[5].

In this situation, physical link is provided, and interconnection point is defined as connection point to the network. The main difference is signalling protocol. In PSTN networks SS7 is mostly used, meaning separate channel for signalling is assigned. In IP-based networks SIP, or maybe some other signalling protocol, is used to transmit data packets with necessary information for call session establishment between end-users. Signalling protocol may not be SIP, but at the moment in most EU countries SIP is mainly used.

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[5] Other less popular signalling protocols can be used in PSTN or IP-based networks case. Schematically there is no need to change anything other than to define the signalling protocol used.
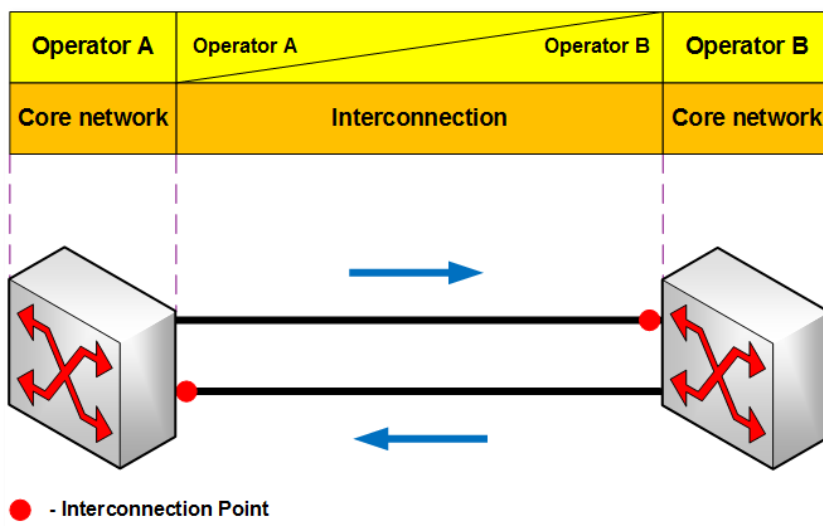
Figure 9: Direct interconnection link

In the case of migration from PSTN to IP-based networks, the existing physical links can be used with only a change of some of the end equipment necessary to connect to the operator network. Technically it is not a problem to provide IP interconnection even if the operators' telephone networks are based on PSTN. In this case a PSTN-IP gateway can be used. However the PSTN operator would incur additional costs to ensure IP interconnection.

7.2.1.2 Interconnection using public IP environment

Interconnection is achieved using the public IP environment, usually Internet, and this approach is becoming more frequent in many European countries to ensure both local and interstate interconnections. This type of interconnection is virtual and costs less than a direct link. An interconnection point in this case could be defined as any public Internet point. Usually the interconnection point is defined at the operators' network connection point to the public Internet. Interconnection using the public Internet is illustrated in Figure 10 below.

Security is a major problem with this type of interconnection. As the IP packets are transmitted in what is essentially a public environment additional security measures should be applied.

Figure 10: Interconnection link using Public Internet

In many countries, regulation is technologically neutral so the method of interconnection adopted is not specified. Therefore, specific problems may arise when using the public Internet, especially in the case of transit.

7.2.1.3   Transit
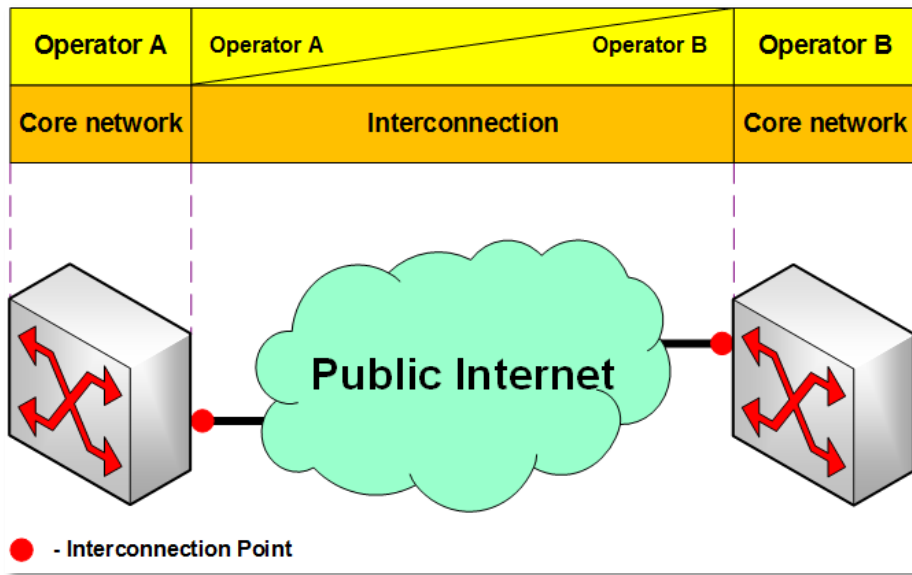
Transit is not an interconnection type but rather a service provided using more than one interconnection. In the BEREC report it is mentioned that almost all operators surveyed[6] connect their networks with the network of the interconnection partner using a direct physical link notwithstanding that (as already mentioned) this is not the only possible interconnection option.

Transit is necessary if there is no direct interconnection between operators. As seen in Figure 11 below and as described earlier, transit may be ensured by using physical and/or virtual interconnections between operators' networks. It is possible that to transmit voice signals from one operator network to another operator network more than one transit service operator will be necessary. Therefore, more than one interconnection will be established in one session to transmit voice in this case. Additionally it is possible for interconnections to be of different types. The probability of voice call service quality parameters degrading is higher in cases where multiple interconnections are used for a single voice transfer session.

Regulatory requirements do not apply to transit services in almost all EU Member States. Therefore it is essential, especially for smaller operators providing both voice services and Internet access services, to separate voice traffic from overall data traffic.

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[6] Only one operator reported that interconnection was achieved using a different approach.
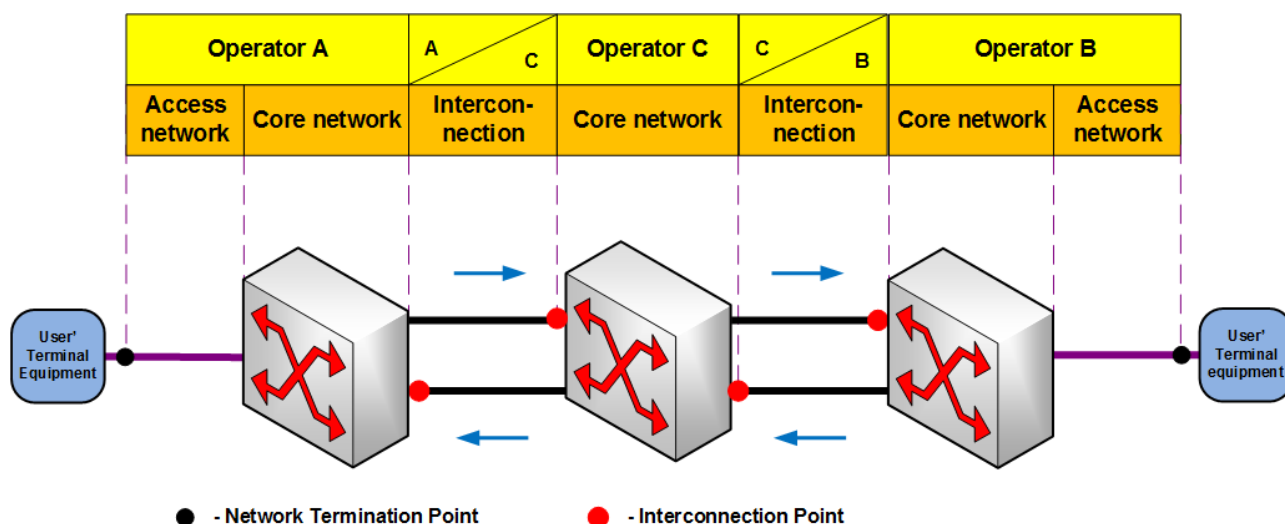
Figure 11: Operators connection using transit

7.2.2 Signalling protocols

In PSTN, the SS7 signalling protocol is typically used, but exceptions are possible. The migration to IP-based networks requires operators to seek new signalling protocols that enable communication between users over IP-based networks.

SIP is the most common signalling protocol used by most network operators for IP-based networks. It should be noted however that there exist several different SIP extensions which are not all compatible in the case of interconnection among operators using different SIP extensions. The BEREC report focuses on SIP as the main IP-based network signalling protocol.

As long as the PSTN exists alongside IP-based networks it is necessary to provide ways to transmit voice even if part of the network is PSTN-based and part IP-based. It should also be considered that operators, that already have an existing PSTN network, will most likely first migrate their core network to IP as already mentioned earlier in this report. But even in this case, there may be operators whose core network is still PSTN-based. Furthermore, the access network, or some parts of it, may still be PSTN-based so as to maintain services for clients using older equipment who may not want to switch to new equipment.

For new operators it is possible, and most of them have availed of the opportunity, to deploy IP-based network technology from the start. In such cases, the access and core networks are IP-based from the beginning. There is no need to ensure PSTN integration in an IP-based core network unless such a network has to interconnect with core networks of other operators which only support TDMvIC if the migration to an IP-based network migration has not been completed.

One of the main problems that arises in interconnection is the signalling protocol capability between two operators. Additional expenses may be incurred from one of the operators to establish interconnection to ensure compatibility between the signalling protocols used on both networks. In this context, we are speaking about the core network which may be PSTN-based or IP-based.

SIP should be the main focus among the different signalling protocols, as mentioned in the BEREC report, because it is the most commonly used protocol by operators in Europe. It is also one of the newest and currently the easiest signalling protocol to use for the transmission of IP-based voice services.

The basic SIP specification is detailed in IETF RFC 3261 [24]. SIP is one of the components of a set that consists of many protocols, that should be used together to establish a complete multimedia architecture. SIP can, and should, be used with other protocols, but SIP itself is a very important part of the set that does not depend on any other protocols and can be used independently. SIP consists of different primitives and can be looked upon as a protocol suite. It is meant for session establishment, session modification and termination. SIP additionally provides a suite of security services, such as denial-of-service prevention,

authentication, integrity protection, encryption and privacy services. SIP can also be used to initiate a session that uses other conference control protocols. Some supplementary services such as call forwarding and caller line identification can be provided.

SIP is widely used to support call connection between soft switches. For this purpose there are two possible routing schemes. The first routing scheme is based on ISDN User protocol (ISUP) encapsulation into SIP messages, where additional information is inserted in the SIP message body. The second routing scheme uses an ENUM-like mechanism to route calls. In the second case ISUP needs to be modified because the call control of the call server does not provide number and route analysis functions. As a result, additional modifications are needed for some network control modes also.

Unfortunately some basic PSTN functions, such as PSTN encapsulation, supplementary service provisioning and others are disabled by SIP. Therefore, SIP extensions are necessary.

There are two possible solutions for interworking between SIP and PSTN. One of them is the SIP for Telephones (SIP-T) protocol suite provided by the Internet Engineering Task Force (IETF) and the other is the SIP with Encapsulated ISUP (SIP-I) protocol suite provided by the ITU-T.

SIP-T is an IETF product, as is SIP itself. SIP-T allows for ISUP signalling to be carried in SIP messages in cases of SIP-PSTN calls, PSTN-SIP calls and PSTN-PSTN calls in IP-based networks. It should be taken into account that the IETF recommendation RFC 3372 [25] notes that SIP-T security is not very high and using SIP-T should only be considered in pre-existing trusted relationships between operators. For SIP-T additional security measures may be needed. It is also mentioned in RFC 3372 that some problems with carrier signalling bridging with end-user signalling may arise. Although the standards provide information about encapsulation and mapping, supplementary service provisioning is not within the scope of the standard. Therefore, the SIP-T specification does not describe any supplementary services provisioning. SIP-T is widely used in inter-soft switch communication as a signalling protocol and some operators already refer to SIP-T as being a legacy protocol. Nevertheless, there are still some operators that use SIP-T actively.

SIP-I is a SIP extension provided by the ITU. SIP-I uses IETF specifications for SIP as a base, and provides additional specifications for SIP-ISUP interworking. SIP-I standards describe interworking between SIP and ISUP/BICC, including general principles, interworking architectures, security, supported supplementary services, etc. It should be noted that the SIP-I specification describes supplementary services provisioning, that is not specified in SIP-T. Interworking models are defined in SIP, SIP-I and 3GPP SIP cases. SIP-I can result in better performance than SIP-T when used as the interworking protocol between soft switches and the PSTN. For bigger telecommunication companies SIP-I is a preferable choice. In some countries, for example Italy, the use of SIP-I is a regulatory requirement and must be used by all operators for interconnection. SIP-I basically includes SIP-T. Therefore, if interconnection is needed, operators that use SIP-I can provide it without incurring additional costs. On the other hand, interconnection from SIP-I to SIP-T is not easily compatible.

3GPP defines a specific extensions for SIP that are necessary in mobile networks as additional specifications are necessary because mobile network structure, protocols and signalling information may be different from fixed networks. 3GPP extensions for SIP can also be used for other cases.

As already mentioned, SIP is the most widely used signalling protocol for interconnection at the moment and it is likely that this will continue in the future. However, the use of SIP is not the only possible way to establish signalling for telephone calls. H.323 is also quite a popular protocol suite for signalling that can be used to establish interconnection between the PSTN and IP-based networks, between two IP-based networks or between two PSTN networks. ITU-T Recommendation H.323 [26] dates from around the same time as SIP. The recommendation is highly detailed and includes information on supplementary services provisioning, tunnelling of non-H.323 signalling messages, specific codecs to be used for fax transmission, audio codecs for voice capability and other aspects of packet-based multimedia communication systems. As mentioned in the recommendation, support of supplementary services is optional. The preferred codec for fax transmission is T.38, and as obligatory audio codec G.711 or G.729, for low bit rate segments, is defined. Other codecs may be used for ensuring voice capability. The recommendation provides a very detailed description of tunnelling of non-H.323 signalling messages that may be QSIG, ISUP, ISDN, DSSI or others. Special activities for Quality of Service (QoS) parameters improvement are considered and mentioned in H.323 and related recommendations. So the H.323 protocol suite description is quite complex and consists of large

amounts of information and improvements are made all the time. Furthermore, H.323 is relatively expensive as its complex nature can be difficult to support in practice. The number of specialists with expertise and experience in H.323 is not so high and additional training on H.323 support is required by network maintenance engineers.

CISCO tried to use SCCP widely and make it more popular, but at the moment the latest CISCO equipment uses SIP. SCCP is used mostly in internal networks that are not needed to be compatible with other networks.

MGCP and Megaco/H.248 are also other signalling protocols that can be used. Both these signalling protocols can work as support for SIP or H.323 or provide signalling as it was originally developed as an easier alternative to H.323. However, these protocols are not typically used for interconnections.

Some operators refer to protocols other than SIP as legacy. While these legacy protocols may be good, they are considered too expensive, too difficult or just not as good as SIP to use. In case of IP-based network interconnection the only demand is for signalling protocols to be the same for both operators at interconnection point. Table 1 below provides an overview of the various protocols discussed in this section and their respective references.

Table 1: Signalling protocols and recommendations

Signalling Protocol	Recommendations	Notes
SIP	IETF RFC3261, RFC3398, RFC2833, RFC3264 and related (RFC3265, RFC3853, RFC4320 etc.)	-
SIP-T	IETF RFC3372, RFC2976, RFC3204, RFC3398 and related	SIP protocol extension
SIP-I	TRQ.2815, Q.1912.5 Profile C	SIP protocol extension
3GPP extensions	ETSI TS 124.229 IETF RFC4083 RFC3455 3GPP TS 24.229 and related (3GPP TS 23.228 and TS 24.228 and TS 23.218)	SIP protocol additional extensions
H.323	ITU H.225, H.323, H.245, H.235, H.450.x and related	-
Megaco/H.248	H.248, IETF RFC3525, IETF RFC3550 and related	Can be complementary to H.323 and SIP.
MGCP	IETF RFC 3435, RFC3550, RFC2705	Can be complementary to H.323 and SIP
SCCP (Skinny)	CISCO documents about SCCP usage	CISCO defined signalling protocol

When operators are in the process of negotiating interconnection, difficulties can arise regarding which protocols to use. One operator may be using one signalling protocol, and other telephone network operator may be using a different signalling protocol. This can make negotiations quite difficult in cases where different PSTN signalling protocols and IP signalling protocols are used to enable interconnection. As described above, there are different ways of achieving interconnection using different signalling protocols but the most effective way is for both operators to agree to use the same signalling protocols.

In order to avoid disagreements during interconnection negotiations, there are a number of options:

▪ NRAs could intervene by defining the signalling protocol to be used in all interconnections (Regulation policy influence);

- Operators could agree between themselves on a protocol that could satisfy both sides (Agreement between operators);
- Combination of the above-mentioned options.

The popularity of SIP is high in Europe among telecommunication service providers and equipment manufacturers and other signalling protocols are not so frequently used. The following sections of this chapter are predominantly based on the assumption that SIP (and its extensions) is the most preferred and used protocol in Europe. Issues relating to the use of E.164 numbers in the SIP environment are dealt with in Section 7.2.7.1.

7.2.3   IP codecs

In the context of this ECC Report, only audio and fax codecs require analysis. Video codecs are not discussed as the main focus is only on voice transmission services.

Before every call session, it is necessary to find agreement about which codec to be used for the session. Transcoding is also possible but quality will most likely be reduced and additional equipment and/or software is required.

The H.323 specification describes minimal requirements for codecs and exactly which codecs should be used in different cases, as already mentioned G.711 and G.729 whereas the SIP specification in conjunction with Session Description Protocol (SDP) does not specify codecs. Therefore, practically every codec could be used if both interconnection sides agree on the one audio codec for a specific call session.

Some codecs require a license for commercial use. This means that, even if hardware is capable of supporting these codecs, additional costs will be incurred for a license. Larger operators can purchase more licenses, but in the case of smaller network operators it is most likely that they will choose only a few codecs, or even just one. So if two smaller operators make an interconnection and they do not have the same codec for selection, it may cause problems and disagreements. Either one operator will have to buy license or transcoding will be required that may cause QoS degradation.

Each operator, if it has more than one usable codec, may choose a codec with a higher priority to be selected to provide the best possible voice transmission. Codecs should be chosen considering different parameters, such as bandwidth, sample interval, bit rate, necessary Mean Opinion Score (MOS) etc. These parameters' values may differ slightly depending on the different platforms and signalling protocols used although the values should be very close in each case. Codecs should also be chosen which correspond with transmission systems parameters. These include available bandwidth, QoS requirements, capacity, memory requirements etc. Some codecs parameters can be seen in the Table 2 which are based on the BEREC report identifying codecs that are more commonly used in Europe.

Table 2: Different codec average parameters (values may differ depending of the transmission system used) [CISCO "Voice Over IP – Per Call Bandwidth Consumption" Apr 13, 2016, Document ID: 7934]

Codec Information				IP Bandwidth Calculations			
Codec	Bit Rate (Kbps)	Sampling rate	Nominal Mean Opinion Score (MOS)	Voice Payload Size (Bytes)	Voice Payload Size (ms)	Packets Per Second (PPS)	Bandwidth (Kbps)[7]
G.711 A-law	64	8000	4.2	160	20	50	87
G.729a	8	8000	3.91	20	20	50	30
G.722	64	16000	3.91	160	20	50	87
GSM-EFR	12.2	8000	4.16	-	-	-	32
AMR-NB	4.75-12.2	8000	varies	-	-	-	varies

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[7] Ethernet

G.711 is the most frequently used A-law audio codec by network operators in Europe. It is also supported by most equipment vendors as it is indigenous to E1 standard. G.729 is also popular while G.722 is becoming increasingly popular. There are other codecs, not listed in Table 2, which can be used. For example G.723, G.726 and others.

Support for fax transmission is still required for telephone networks today. This can be achieved in different ways. Firstly, an audio codec can be used which provides the possibility to transmit different kinds of data. G.711 supports fax transmission but it is not specified during transmission that fax is transmitted. Secondly, it is possible to use a codec that is strictly meant for fax transmission. The most popular of these is T.38 which was exclusively developed for fax transmission. T.38 is currently the most widely used codec for this purpose by the different operators in Europe.

From an operator perspective, it is possible to establish interconnection using the following solutions:

- Ensure that on both sides of the interconnection that at least one codec is supported;
- If different codecs are used then transcoding is needed which can result in QoS degradation. Nevertheless it is considered a viable solution if the QoS degradation is acceptable for both operators.

From a regulatory perspective, it is important to minimize the possibility of disputes between operators when issues arise about codecs and network codec compatibility. It is possible to define specific codecs for voice transmission and make those obligatory for all telephone service operators in the regulator's jurisdiction.

7.2.4 Points of interconnection

Determining the exact point of interconnection in the case of direct interconnection between two networks using the SIP protocol could be defined in the same way as in case of PSTN interconnection using the SS7 signalling protocol. This means that the point of interconnection can be specified in geographic location.

In case of virtual interconnection via public internet, the point of interconnection determination becomes more difficult and is a current issue for discussion. With IP interconnection via public internet case equipment can be moved and interconnection can be basically defined in any geographic location where the equipment is at that moment in time. Usually, most virtual interconnection points are defined as the location where the connection point to the public internet is made. To be consistent with the existing interconnection scheme, this is an appropriate approach for migration period because it can be used for both traditional and IP-based interconnection. It may also be important to specify an exact geographical location in such cases thereby allowing operators to know the exact location of the interconnection equipment.

7.2.5 Quality of service

There are no specific Quality of Service (QoS) requirements and measurement parameters for voice telephony interconnection. However, periodical end-to-end off-net QoS measurements on regular basis may be stipulated by national legislation. In such cases, the QoS parameters are the same as for on-net voice telephony. It should also be noted that these parameters do not only apply to the interconnection QoS because the end-to-end measurement results include interconnection QoS only as a part of the overall measured loop and the influence from the operator's interconnected equipment and access network, while significant, may not normally be enough for situation monitoring.

QoS degradation may be identified by analysing the measurement results for voice telephony end-to-end off-net. This will allow the direct IP-based interconnection QoS control to find the cause of the QoS level degradation. Using this approach, the parameters for measuring QoS of Internet access services (addressed in Section 6.2.2) are applicable. These measurements should focus only on interconnection and should not cover customer aspects for voice telephony services. It should be noted that such measurements should only be made in cases where the interconnection link (physical or virtual) is disconnected from the operator's equipment. Therefore it will be difficult to arrange such measurements on regular basis.

7.2.6   Provision of supplementary services

The provision of supplementary services is an important aspect to consider when discussing interconnection for networks providing voice telephony services.

Some signalling protocol specifications describe the provision of supplementary services or mention the necessary requirements for it. In the context of interconnection, the majority of supplementary services are an optional requirement. Technical standards do not oblige voice service providers to provide supplementary services. As defined in ITU Q.1912.5, SIP profiles can provide specific supplementary services if needed, and in SIP-I most services are already supported by encapsulation. H.323 describes the provision of supplementary services but notes that it is not an obligatory demand for this signalling protocol.

As already mentioned, in the case of SIP-I some supplementary services are already part of ISDN encapsulation mechanism, so there is no need for any additional functions. Supplementary services that are part of encapsulation mechanism include the following:

- Calling Line Identification Presentation (CLIP);
- Calling Line Identification Restriction (CLIR);
- Connected Line Identification Presentation (COLP);
- Connected line Identification Restriction (COLR);
- CLIP no screening;
- COLP no screening;
- Direct-Dialling-In (DDI);
- Call Deflection during alerting (CD);
- Call Forwarding (CF);
- Explicit Call Transfer (ECT);
- Anonymous Call Rejection (ACR);
- Reject Forward call (only if Call Forwarding indication is provided by ISUP);
- Call waiting (CW);
- Three-Party conference (3PTY) (depending on special situation via destination IP-network);
- Closed user Group (CUG);
- User to user signalling (UUS).

For Malicious Call Identification (MCID) the required parameters can be taken from encapsulated ISUP MIME, but the IP bearer cannot be held after the release of the call. Call Hold (CH) interworking to SIP-I is achieved via the encapsulated CPG message without the need for additional interworking. Some other supplementary services also have specific conditions of use and a full description can be found in ITU Q.1912.5.

The requirements for the interconnection interface should be to exchange the information to allow the operation of supplementary services. The main problem is that even if call originating network provides supplementary services, there is no guarantee that call terminating network can support these supplementary services. This may happen in cases where different signalling protocols or SIP extensions are used, or the terminating operator sees no need to provide these services. From a regulatory perspective, some basic supplementary services which are expected by customers should be provided. These will need to be defined in national legislation.

7.2.7   Impact of migration on existing telephone numbering ranges

In November 2012, the ECC's Working Group Numbering and Networks (WG NaN) published a Green Paper [28] on the future of numbering entitled "Long Term Evolution in Numbering, Naming and Addressing 2012 – 2022". The Green Paper's main conclusion was that "*E.164 numbers will still be the most common universal identifiers used for the provision of electronic communication services which will have to be coordinated at the national and international level (ITU-T). The main function will be naming, and the national public authorities will have to manage the resource in the interest of all stakeholders*".

7.2.7.1   *Impact on numbering and routing*

Traditionally telephone numbers were used to route calls and to identify physical access paths. As networks have evolved this function has also evolved. As the migration from PSTN/ISDN to all-IP continues telephone

numbers will, as the Green Paper concluded, be used as resolvable names where origination and termination points will be identified using IP addressing. This change in functionality will be seamless for users who will still be able to make and receive voice calls and messages using existing E.164 numbering resources. Therefore the impact on numbering for users as a result of PSTN/ISDN migration to all-IP is not likely to have significant regulatory implications.

The migration to all-IP will have an impact on routing regarding the way telephone numbers are expressed and provisioned on networks. In an all-IP environment, where the SIP protocol is used, it is important to distinguish between "locally unique" and "globally unique" numbers. Locally unique numbers are unique only within a certain geographical area or a certain part of the telephone network e.g. a PBX, state or province, a local exchange or a particular country. PSTN networks are organised to handle these differences and can route calls where numbers are dialled using either a local, national or international digit string. In IP-based networks, this may not always be the case and a common approach would be beneficial.

According to IETF RFC 3966 [29], globally unique numbers are identified by the leading character and the digit string is composed using the "+" character followed by the country code (CC) and national significant numbers (NSN) as specified in ITU-T Recommendations E.123 [30] and E.164 [31]. Globally unique numbers are unambiguous everywhere in the world and are therefore most suitable for use in IP-based networks. Before making any recommendations in this regard, it will be necessary to examine the impact this may have when dealing with maximum telephone numbering lengths. ITU-T Recommendation E.164 defines a maximum number length 15 digits[8]. Many national administrations are introducing numbering ranges for M2M/IoT services which have a longer format than those typically used for interpersonal communications. Furthermore, telephone numbers are also often prefixed with number portability routing codes (typically ranging from 2-7 digits) in order to efficiently route calls to the terminating operator. The impact on routing of using numbers with a total length of more than 15 digits needs to be carefully evaluated.

7.2.7.2 *Impact on number portability*

Number Portability (NP) has been a key competition enabler since electronic communications markets were liberalised. NP solutions have enabled end-users to switch service providers within the national market while retaining their telephone number. These solutions are based on standard technical solutions (usually NP mechanisms defined by ITU-T and ETSI on traditional telephony network technology), nationally adapted and implemented with country-specific characteristics. An IP-based network infrastructure would facilitate the use of alternative mechanisms for mapping E.164 numbers to URIs in order to identify the terminating network serving these E.164 numbers. For example ENUM-like mechanisms [32] could be used inside DNS systems for routing purposes. Large investments have already been made by the industry stakeholders in order to meet their respective number portability obligations and any future migration to ENUM-like mechanisms should take into account existing national investments thereby minimising the impact on current number portability systems, processes and procedures.

7.2.8 Network security

Network security is more difficult to ensure in the IP-world than in the PSTN/ISDN. However, security breaches and severe outages are also known to have happened in PSTN networks due to problems with the signalling system (SS7). In the "open" or public Internet, security is often widely questioned due to severe incidents with piracy, sabotage and denial of service attacks. Thus many operators are reluctant to rely on the public Internet for interconnection with networks of their partners. Instead, they prefer direct IP interconnection thereby reducing the risks associated with the public Internet.

Operators apply different measures and mechanisms to ensure the security of the IP interconnection:

- Topology hiding, a function which allows the hiding of network element addresses from 3rd parties and obscuring the architectural layout of those elements;
- Use of Session Border Controllers;
- Encryption by using, for example, IPSec;
- Authentication of the connecting parties, particularly at BGC;
- Access Control Lists: Filtering the packets and allowing only matching traffic to be forwarded;

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[8] 15 digits in total, excluding the international prefix ("+", 00, 001 or 011 etc.) but including the country code.

- Reverse Path Filters: Filtering incoming traffic to ensure the traffic received is limited to that received from IP addresses that are sent via that interface;
- Traffic Policing: Controlling the rate of incoming or outgoing packets or requests;
- Application Level Relaying: Performed by terminating a particular application request session on one side of the relaying device and then relaying the request/session to another network element;
- Deep Packet Inspection: Providing the ability to look into the payload that is carried by the packet and use the contents to perform filtering or rate control;
- Secure RTP: A protocol that encrypts RTP media packets and provides authentication and integrity for those packets;
- DNS Security: Providing an additional layer of security for DNS clients;
- Media Filtering (Pinholing): A technique filtering RTP protocol packets;
- Use of Firewalls;
- Introduction Detection Systems: Devices or software detecting unauthorized access to network resources;
- Device Hardening: To ensure elements are less vulnerable to network intrusions;
- Logging and auditing: A basic security practice;
- IP addresses hiding.

8   TIME SCHEDULE – PLANS

Considering the diversity and complexity of telecommunications infrastructure and markets in the various European countries, it is obvious that the migration from PSTN/ISDN to IP-based networks cannot be concerted and synchronised in all countries. Consequently, each operator (incumbent or competitor) will have its own phase-out timing depending on a number of different factors considered in the development of its migration strategy. In November 2014, the ECC's Project Team Technical Regulatory Issues (PT TRIS) conducted a survey with CEPT member countries to determine among other things:

▪ if the incumbent fixed network operators have an official plan for PSTN migration;
▪ and if yes, what is the time schedule for expected shut-down of PSTN-services.

The responses to the survey are contained in Annex 2 to this ECC Report[9]. The results show that the incumbent has a migration strategy in a majority of countries and that in many cases the process has already started. For those countries still having no migration strategy it is expected that the main drivers for migration (discussed in Chapter 3) will lead them to initiate a migration strategy to all-IP in the not too distant future.

The results of the ECC survey, as well as the BEREC report "Case Studies on Migration from POTS/ISDN to IP on the Subscriber Access Line in Europe" [33], show that the speed of migration varies widely between countries and operators. While migration is already completed in some countries (Macedonia, Slovakia, Croatia, and Montenegro) since the end of 2015, others are considering 2025 as the target deadline. This significant gap can be attributed to various factors which are worth examining.

8.1   DIFFERENT MIGRATION SCENARIOS

The selected migration scenario (see Chapter 5) can influence the total duration of the migration project to an IP-based network. The ideal solution would be to set up a complete IP-based network (core and access) in parallel with the existing PSTN network continuing to operate. Then, very quickly, migrate the users onto the new network before decommissioning the old network. If the implementation timeframe for the core network is relatively short, the construction of a new access network can be dependent on many parameters as discussed hereafter. Once the construction of a new access network is complete, moving customers to the new network could then be very fast. The results to both the ECC survey and the BEREC report demonstrate that the implementation of a complete IP-based network in parallel to an active PSTN network is an ideal scenario but one which is not considered by the vast majority of operators. More realistic scenarios are given priority including a temporary solution, full PSTN / ISDN migration or a combined approach. These solutions require more implementation time but a phased migration can easily be planned and controlled.

8.2   CORE NETWORK AND ACCESS NETWORK

Generally speaking, one can assume that a core network based on all-IP requires significantly less network infrastructure when compared to a PSTN core network which is very hierarchical by design. The whole transit network, and its fully IP-based components, are simplified and require fewer network nodes to be implemented. An IP-based network includes, in general, two IMS cores installed at two separate locations to ensure proper redundancy and to guarantee the safety of the entire network, as well as a number of gateways (access, signalling, and trunking media) depending on the services offered and the number of end-users.

The reduced number of elements required for an IMS core network, and its peripheral components, can therefore easily be implemented at locations already used by the transit network of the traditional PSTN/ISDN network. Furthermore, these new network elements require a lot less resources for floor space, power and cooling. Only the implementation of fast high capacity and extremely reliable links may require

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[9] Annex 2 is published as a separate document to this ECC Report entitled ECC Report 265 – Annex 2.

consequent works. The implementation timeframe for a complete IMS core network (which can typically be completed within one year) can be considered as short in this context.

The situation is completely different with regard to the access network. While some operators favour the migration of the subscriber access line (SAL) from PSTN/ISDN to IP, others will continue to use PSTN/ISDN access by converting to IP in the Multi-Service Access Node (MSAN) or Access Media Gateway. Some operators also consider a mix of the two approaches. The first solution (IP access line) may require a replacement of the traditional copper line with optical fibre. This process can take a lot of time if the network is particularly dense (e.g. urban areas, cities) or if the topography is challenging (e.g. rural or mountainous areas). For the MSAN solution, each subscriber line requires specific equipment whose implementation time can be significant. Thus, the implementation of the new access part of an IP-based network depends on these various factors and can vary significantly from one operator to another (typically requiring between 1 and 5 years, or even longer in some cases).

8.3 STATE OF THE INCUMBENTS AND OTHER OPERATORS

The position of the incumbent and other operators in a particular country is a significant factor that can influence the migration to IP-based networks. The planning depends greatly on the size and number of operators in terms of market share, number of customers, geographic coverage and the ownership structure of both large and small operators in the market (i.e. independent company or subsidiary of an international parent group company). The planning should also take into account the market structure in terms of the regulated wholesale products that are available to other operators.

Network operators planning migration projects should take into account any potential impacts on both their wholesale and retail customers and they should clearly communicate their migration plans and allow sufficient time for customers to prepare for the migration. In the case of wholesale customers, the migration plans should take account of investments already made and, where necessary, provide alternative equivalent wholesale products. These measures are necessary for ongoing competition and protection of end-user rights.

A fully independent operator will have to formulate its migration strategy by counting only on its own resources, from planning to implementation. However, IMS platform and core network providers with a multi-country presence (of which there are only a few big international payers) have developed extensive experience in the implementation of such platforms around the world and can therefore facilitate the planning and the active phases of the roll-out. However, for the access part, independent operators will be very dependent on their strategic choices and scenarios.

The migration strategies of incumbents and other operators who are subsidiaries of a parent group company may depend on the parent company's priorities and migration activities by its other subsidiaries in other countries. It is possible that they can benefit from economies of scale and synergies to enable a faster and less costly rollout. Indeed the first subsidiary operators of the parent group company to be migrated to IP-based networks can be regarded as pioneers and the experiences gained during these migrations can help to learn from mistakes made and to optimize the migration strategies for other subsidiaries.

The German incumbent Deutsche Telekom demonstrates this point perfectly. It has subsidiary operations in several European countries. Deutsche Telekom migrated to all-IP in Macedonia, Slovakia, Croatia and Montenegro in 2015 and is committed to shutting down its other European PSTN/ISDN networks by the end of 2018 with target shutdown dates provided for the following countries.

- Hungary   2016;
- Romania   2018;
- Greece   2018;
- Germany   2018.

According to Claudia Nemat, Member of the Deutsche Telekom AG Board of Management for Europe and Technology [33], *"Within the Group, we have been switching up to 100,000 lines a week – which is the highest migration rate in Europe."* Deutsche Telekom's migration strategy has entailed the first migrations taking place in relatively small countries in terms of surface area and number of customers. This enabled

Deutsche Telekom to learn from its experiences in smaller markets before considering migrations in larger markets.

8.4 NUMBER AND TYPE OF CUSTOMERS

The rollout of the access network is of course dependent on the number of subscriber lines that need to be migrated. A very large network with many customers will require a longer migration timeframe than a network covering a smaller area with fewer customers. The concentration of the customers also plays a big role. Densely populated urban areas are normally migrated more quickly than rural areas where the distance between the MSAN and the customer can be a problem which requires modification of the equipment.

Larger companies and their corporate networks as well as industrial areas are also more quickly migrated onto IP-based networks when compared to residential areas where the topography is more challenging (e.g. rural and mountainous areas) where access to transmission resources are more difficult to account for in new IP-based networks.

The average migration time by customer is thus variable, but for example the Swiss incumbent (Swisscom) gives a time of 15 minutes (interruption at the customer) per line when the migration is well prepared on both operator and customer side.

8.5 NETWORK GRANULARITY

The tree structure of the access networks to be migrated has some influence on the migration time. Some operators, especially the incumbents, historically developed very complex networks whose terminal exchanges were numerous and very decentralised. This high granularity allowed for a dense coverage in all the territory considered but which can now act as a brake when it comes to migration to an IP-based access network. As a rule, the greater the granularity of the network the longer the migration timeframe.

9   POTENTIAL CHALLENGES

In most European countries, announcements about forthcoming migration projects from PSTN/ISDN to IP-based networks have generated a lot of debate, and indeed emotion, regarding the potential implications for end-users. Such debate is inherent to such evolutionary developments but is also not without basis. This chapter discusses some practical problems (such as the fundamental powering of end-user equipment) and may give guidelines to regulators confronted with end-user demands and expectations.

9.1   ISSUES REGARDING POWERING

Different options and scenarios exist for powering the set-up boxes. There are two options where the set-up box (e.g. modem and/or decoder and/or router and/or gateway) may be installed.

- **Option 1** (Figures 12 and 13) – The set-up box is installed outside the customer premises (for example, in the staircases or basements of apartment blocks or in the operator' premises e.g. in cable cabinets).
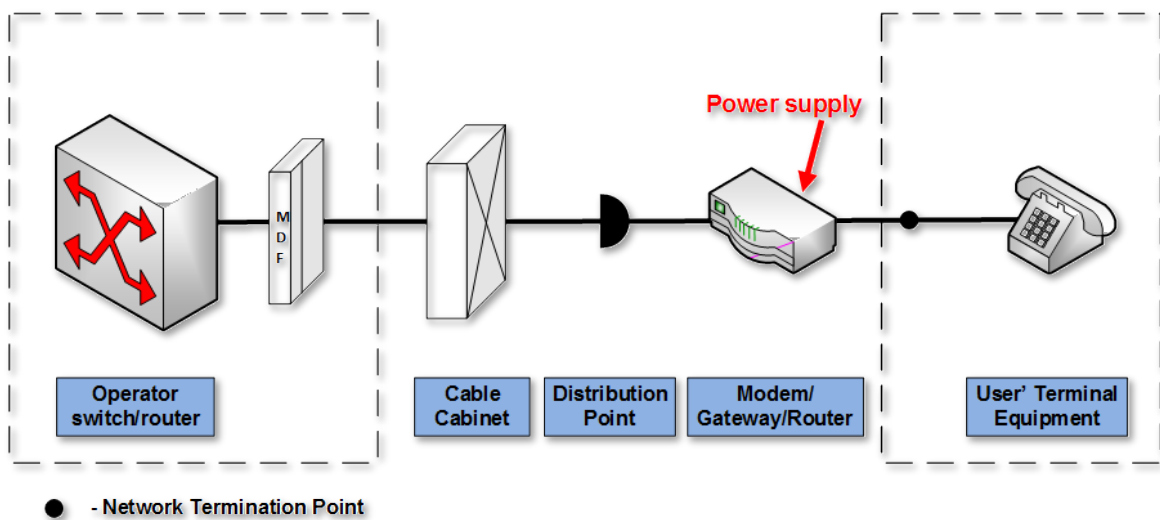

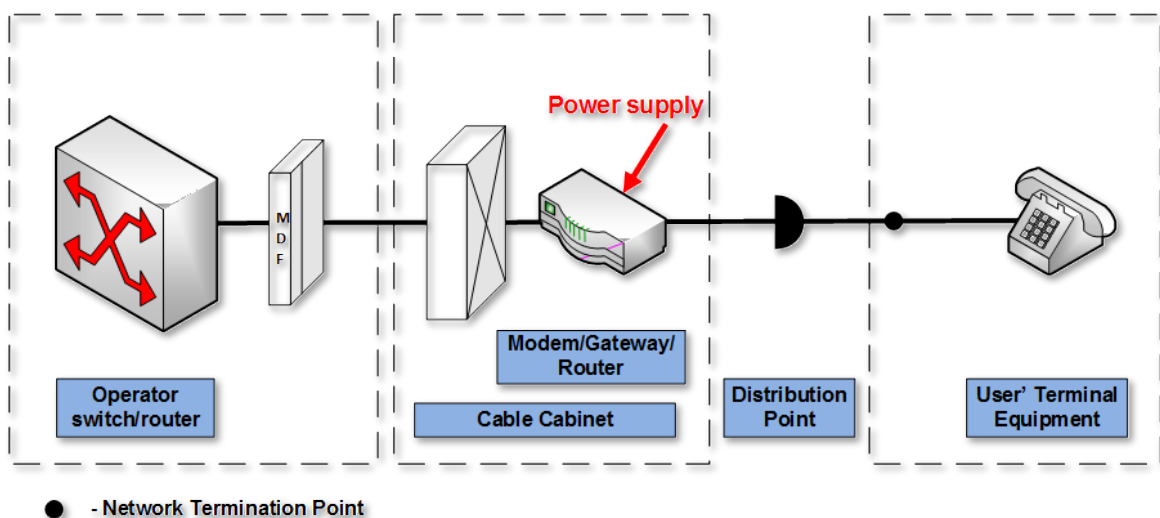**Figure 12: Set-up box outside customer and operator premises**

Figure 13: Set-up box in operator premises

The only difference between Figure 12 and Figure 13 is the place outside the customer premises where the set-up box is installed. In case of Option 1, the operator is responsible for all power supply issues.

- **Option 2** (Figure 14) – The set-up box is installed in the customer premises (for example within the customer's apartment in the case of a private customer). With this option the customer is responsible for providing and guaranteeing power supply if the set-up box does not have battery. The customer also absorbs the costs of electricity.
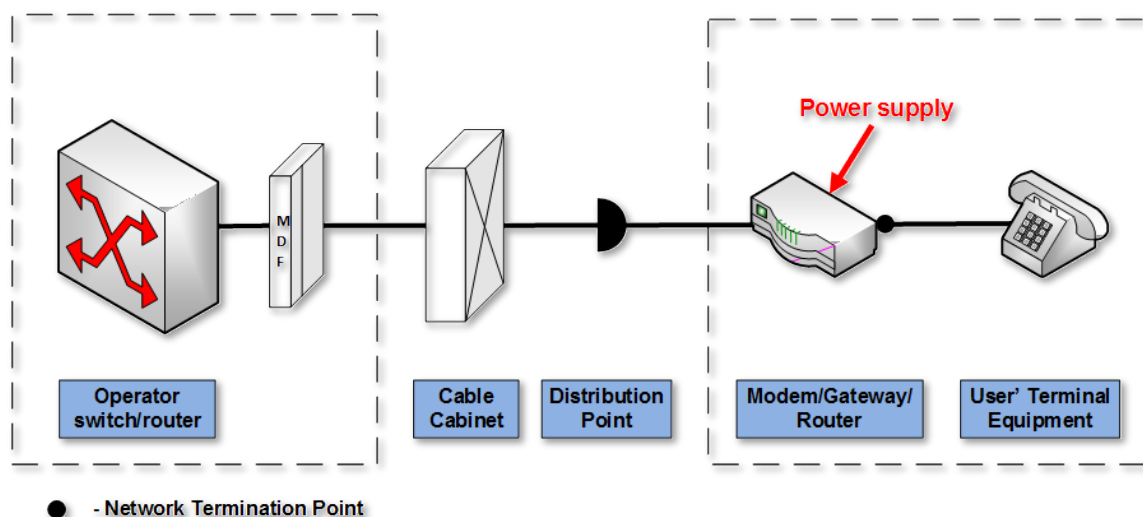

Figure 14: Set-up box belongs to operator and installed in customer premises

With Option 2 all potential problems regarding the powering for the operator's equipment (for example the fuses or other types of required protection equipment to avoid interruptions to power supply because of high voltage or high current) have to be solved by the customer. Otherwise the telecommunication service will not always be available. Additionally in some cases, the operator may ask the customer to install the power outlet for connecting the set-up box to the electrical network or to pay an extra charge for the power outlet installation.

- **Scenario A (for both options)** – Customer has a PSTN-based telephony service together with other services such as access to Internet and IP TV (bundle of services) from the Operator before technical solutions (e.g. ADSL on copper line) will be changed to IP-based solutions. The package of services was ordered by the customer and the set-up box was installed to provide the service. In such cases, the power supply issues are usually included in the customer's contract and the customer agrees to resolving powering problems in his own premises.

  During the migration to IP-based networks, one type of set-up box suitable for one technology may be changed to another one suitable for the IP-based technology. This means that unexpected changes regarding power supply could arise in the case of Scenario A.

- **Scenario B (for both options)** – Customer has only a PSTN-based telephony service provided by copper local loop and does not want to have any additional services such as access to Internet or IP TV. This scenario does not cover ISDN BRA as it is provided using terminal equipment which is part of the operator network. In case of ISDN BRA, Scenario A is applicable.

  Normally there is no problem if Option 1 is used as it does not depend on the customer's power supply and there is a large range of different technical solutions available to provide power supply for the set-up box for this option.

For many years PSTN-based telephony has been provided via a copper local loop. As the power supply was provided by the local exchange there was no need for including provisions in customer contracts regarding power supply. Given these existing expectations, it may be difficult to explain to customers why, after the migration to IP-based networks, they will be responsible for the provision and payment for power supply for equipment which belongs to operator. Furthermore, it may be difficult to explain to customers why it will not be possible to call the emergency service when the power supply is interrupted.

The problem with guaranteed power supply is applicable for Option 2 in both scenarios but for Scenario B the situation is more risky, from a consumer protection point of view, and this problem has to be solved by the operators.

Additionally, it should be noted that some operators provide the set-up box to the customer like a free gift. In this case, as illustrated in Figure 15, the NTP will be placed before the set-up box. This means that the set-up box will not be a part of the network anymore but will be the terminal equipment and the customer will be responsible for all the issues related to the terminal equipment.
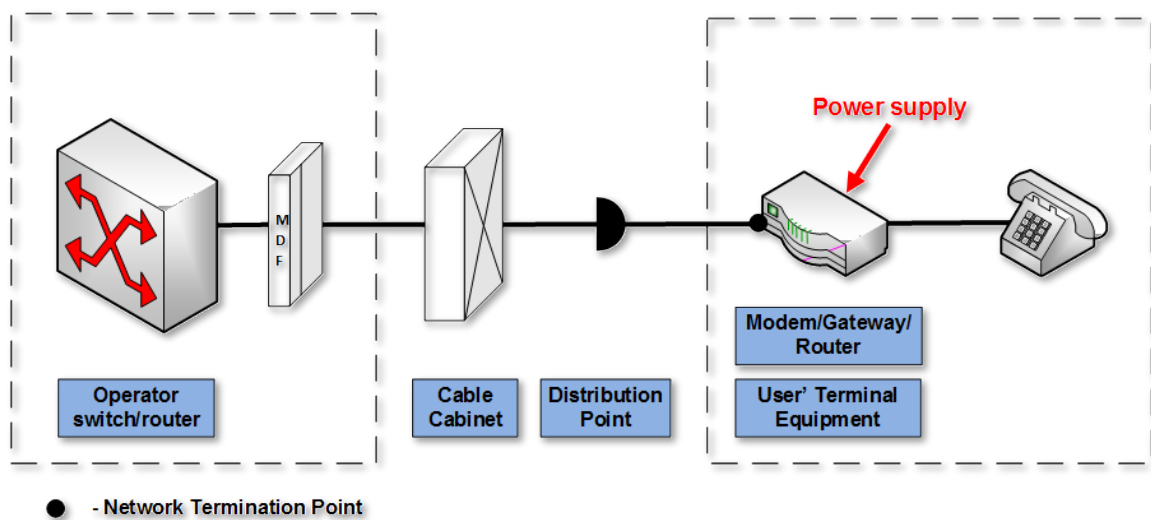

Figure 15: Set-up box belongs to customer and installed in customer premises

9.2 CHANGE OF PARADIGM (RESPONSE TO END-USERS)

The migration to IP-based networks is likely to have a significant sociological impact on users. While not a revolution, it can be considered as a change of paradigm. After decades of "conventional" fixed telecommunications without much evolution, the ubiquity of telecommunications via the public Internet has disrupted the traditional practices of the end-user who may have questions about this sudden announced and compulsory technological change. For some sectors of society, the migration to all-IP represents a transition from a stable and reliable service domain to one which may be considered unknown, vague and unpredictable.

One might argue that younger end-users are likely to be less-impacted. Apart from age, other demographic factors that are influential include level of education, profession, wealth etc. The propensity of customers' willingness to migrate is something that has to be considered by operators who must absolutely reassure those that may have a lower propensity to migrate by communicating, in a simple and understandable manner, the benefits of migration and the changes that will come with it.

Marketing material should be designed with their customers' needs and expectations in mind. Service providers should be able to provide for a seamless migration process and commit to delivering products and services that offer reliability, quality and value for money. Customer communications should be specific in terms of the features of new VoIP telephony and other service offerings based on IP-based networks and the

information provided should be comparable with their competitors' offerings. Regulators will have an important role to play in that regard.

9.3 PRIVATE NETWORKS

9.3.1 Small and medium size private networks

In PSTN, private networks which only have a need for two simultaneous telephone calls can be served by basic rate access to ISDN (ISDN-BRA). Standard VoIP-based services offered via ADSL or VDSL can typically provide two simultaneous telephone calls and can therefore replace ISDN BRAs if the copper twisted pairs are not too long.

In ISDN, private networks requiring larger telephone capacity are supported by multiple ISDN-BRA or by one or more primary rate accesses (ISDN-PRA). One ISDN-PRA is capable of supporting up to 30 simultaneous telephone calls. In VoIP technology these ISDN-PRAs are often replaced by SIP trunks. The capacity of a SIP trunk is limited by the transport capacity of the IP network access (one telephone channel requires about 100 kbps). The SIP Forum's technical recommendation [34] provides a basic set of specifications for SIP trunks.

9.3.2 Large private networks (e.g. corporate networks)

A large private network is typically made up of small individual networks located at geographically distributed sites of one company which are interconnected as one larger network.

PSTN/ISDN corporate networks for telephony consist of small switches at the various remote sites of a company, which handle company internal calls and external calls made to and received from the PSTN. These small switches are connected via PCM Trunk lines where the bigger switch is usually located at the company's main headquarters. The costs for these trunk lines, which are provided by the public network operators as leased lines with a dedicated reserved capacity, are relatively high compared to VPN connections via the public Internet.

Today, the various sites of a company are usually interconnected for data transmission with IP-based networks. These data connections are often also used for telephony services based on VoIP. These VoIP services typically use SIP for Session control. The additional costs incurred by utilising the data connection for VoIP are relatively low as the voice communication typically requires much less bandwidth than many other data services required by a company.

The market demand for corporate networks is gradually moving from PSTN/ISDN to IP-based solutions. In the PSTN/ISDN solutions, calls from the different sites of a company to the PSTN were set up by the small switches in those sites. Those smaller switches often have their own local trunk lines to the PSTN. With IP-based solutions, calls to the PSTN are typically set up by transferring the request first via VPN to the headquarters and via the IP-Switch at the headquarters to the PSTN. This approach may pose some challenges in locating callers and routing calls to the right PSAP in case of emergency calls, similar to the nomadic VoIP problem outlined in section 6.4.2.

9.4 NON-VOICE SERVICES

Burglar alarm systems and home care systems often use modem transmission or DTMF signalling to transmit digital data.

Regarding the various modem technologies used, these systems face similar problems as addressed in the section 9.5, which deals with the migration problems related to analogue fax transmission. The issue with DTMF transmission is addressed in section 9.7.

Voice service providers argue that these kinds of services are not covered by the service they offer, which is voice transmission only. So they consider themselves as not being in charge of supporting these non-voice services and therefore they recommend to "upgrade" old equipment to make it compatible with VoIP or to use alternative access methods such as GSM diallers instead of modems. They also consider the service

provider operating the burglar alarm system or home care systems to be in charge of helping their customers to prepare for the migration from PSTN/ISDN to VoIP.

For IP-based networks, it is of course always recommended (from a technical perspective) to use IP-based data transmission instead of using voice communication channels based on VoIP for modem type data transmission. But an upgrade will create extra costs which will be transferred directly or indirectly to the final customer, who is using the burglar alarm system or home care system.

Elevator alarm systems face the problem that the communication channel for emergency situations has to work in case of power outages. Analogue or ISDN connections provide remote powering for the simple communication systems, which are installed in elevators cabins, but this is not the case with VoIP connections based on DSL or fibre. The elevator industry has prepared itself for a forced migration to VoIP via DSL or fibre by installing a workaround based on mobile terminals. The battery unit ensures operation of the mobile unit when outages of the main power supply for the elevator occur. An elevator vendor will typically transfer the additional costs for this workaround to its customers.

9.5 FAX TRANSMISSION

9.5.1 Technical issues with the fax transmission

The provision of facsimile communication service is a Universal Service obligation. There are still many users (e.g. lawyers) who cannot replace a legal binding fax transmission by an email with attachment. The following paragraphs will address the technical problem with Fax transmission in IP-based networks.

In IP-based networks a bit error leads to a packet loss, because an IP-packet with an incorrect checksum is discarded. A packet loss in a typical VoIP transmission network with 20 ms packetising, creates a signal gap in the region of 20 ms. Such a gap is not a problem for voice communication (it is almost impossible for a human to hear) but in the case of a fax connection such a gap could be severe.

If packet loss appears repeatedly, it could lead to an increase of the jitter buffer size in the equipment which is part of the VoIP connection, e.g. the IAD. This causes, in addition to the 20 ms gaps, a time shift of the received signal, which will very likely cause the demodulation module of the fax equipment to lose synchronisation, with the consequence that the fax transmission aborts.

The longer the duration the fax transmission lasts, the higher the risk that packet losses will appear which increases the probability that the fax transmission will not be completed. Fax users have already raised concerns that the transmission of faxes with a significant number of pages is no longer reliable following migration from PSTN/ISDN to IP-based networks.

In packet-based transmission, a synchronisation between the transmitting and receiving endpoint is, in many cases, no longer required. Jitter buffers compensate for the delay variances between the packets in a transmission session. In a voice or fax transmission over IP, the clock on the transmit side determines the rate at which packets are sent. The clock on the receiving side determines the rate at which the packets are processed and the rate at which the voice or fax samples are processed in the codec. During long transmission periods the difference between those two clocks may cause the jitter buffer to reach its limit. When this happens, packets might get lost and in many cases a time shift in the voice or fax signal may appear. A voice communication is not strongly affected by this but the modem in a fax machine will usually abort the transmission if this occurs. Only terminal equipment which adjusts the processing clock according to the buffer status can cope with this problem.

Issues with different vendors' implementation of MGWs and CPEs to detect fax and modem preamble tones can also be a potential source of interworking issues that require a greater level of interoperability testing by operators than compared with legacy handling of fax and modem.

Useful information about the fax problem can be found at http://www.webtorials.com/content/2012/11/why-fax-over-ip-fails.html.

9.5.2   Technical solutions for the fax problem

ITU-T recommends a fax transmission protocol T.38 to be used in IP-based networks. This protocol was invented to have a pure digital transmission, without modulation and demodulation for an analogue voice channel with a bandwidth of 0.3 to 3.4 kHz. Unfortunately, the utilisation of T.38 in VoIP-based networks creates a lot of compatibility problems (there are different RFCs addressing different version of the T.38 protocol and the way the standard analogue faxes (using T.30 Protocol) interoperate with a T.38 is not fully solved).

The "Fax Over IP Task Group" of the SIP Forum deals with T.38 problems and published a problem Statement paper. The document lists 13 problems and provides recommendations [35].

The i3 Forum together with the SIP Forum worked out a technical specification for Fax over IP [36], which is intended to deliver guidelines for reliable FoIP call setup to help service providers.

As these guidelines are not yet implemented in the networks, simply referring to T.38 cannot yet be considered as an overall solution for the Fax problem in VoIP networks. There is still a lot work to be done.

9.5.3   Recommendation from service providers

Voice service providers offering fax service based on VoIP networks provide some recommendations to their customers who are struggling with fax problems. These mainly advise:

- Selecting a speed of 9600 Baud or less;
- Switching off the error correction mode (ECM);
- Disabling T.38 switch over;
- Not transmitting too many pages in one session.

9.6   QOS ISSUES

VoIP implementations may face problems with latency, packet loss, and jitter. For VoIP to be a replacement for PSTN telephony services, customers need to receive the same quality of voice transmission that they receive with traditional telephony services. This means consistently high-quality voice transmissions. Like other real-time applications, VoIP is extremely "bandwidth-and-delay" sensitive. For VoIP transmissions to be intelligible to the receiver, voice packets should not be dropped, excessively delayed, or suffer jitter.

VoIP can guarantee high-quality voice transmission only if the voice packets, for both the signalling and audio channel, are given priority over other kinds of network traffic. For VoIP to be deployed so that customers receive an acceptable level of voice quality, VoIP traffic must be guaranteed certain compensating bandwidth, latency and jitter requirements. QoS ensures that VoIP voice packets receive the preferential treatment they require. In general, QoS provides better (and more predictable) network service by providing the following features:

- Supporting dedicated bandwidth;
- Improving loss characteristics;
- Avoiding and managing network congestion;
- Shaping network traffic; and
- Setting traffic priorities across the network.

In general, for voice telephony services the quality relates to answering the questions:

- How successful am I in reaching the called party or the requested session?
- How long is the waiting time to get to the called party or the session after the initial set-up request?
- After I succeed in making a voice connection, how satisfactory is the conversation quality or voice quality during the call?"

Defined parameters for the QoS assessment from a customer perspective could include:

- Unsuccessful call ratio;
- Call setup time; and
- Speech transmission quality (MOS/ PESQ/ POLQA).

The set of QoS parameters is designed to be understood by customers. Sub-sets of these parameters can be selected for use in different circumstances or service platforms.

Additionally, for the voice transmission in IP-based networks, the relevant parameters for the QoS assessment in the IP environment could include:

- Delay variation –jitter (download/upload);
- Packet loss ratio (download/upload);
- Packet order (download/upload);
- Packet discards; and
- SIP call flow request times for REGISTER, INVITE, and BYE signalling messages.

The above-mentioned parameters for the IP environment may affect the customer's perception of the voice service quality when measured in terms of the success of call. Waiting time duration and speech transmission will also have an impact on the customer's perception.

For voice-over-IP applications, the variance of latency and associated jitter must be kept to a minimum otherwise call quality will be degraded. High packet loss (for example, more than 5% sustained over a short period) will result in broken sound during calls. The above-mentioned parameters might be affected by high transmission speed fluctuations also. If the connection is regulated in such a way, for example, an operator may supply a 10 Mbit connection by allowing 100 Mbit for 10ms and pause for 90ms. The higher data transmission speed is better, but a quality connection will also demonstrate very little fluctuation in data transmission speeds. The rapid speed changes indicate that data are impacted and the level of quality will be lower. It will definitely affect multimedia applications such as voice services.

From the customer perspective degrading quality of services due to migration towards IP-based networks is related to the possible degradation of some parameters such as call setup time and voice transmission quality. Depending on technical solutions implemented in the operator's networks, the call setup time values in IP-based networks may reach the time values that are typically obtained on mobile networks.

For voice telephony services, call setup time measurements provided in some Latvian operators' networks show the following results:

Average call setup time, seconds

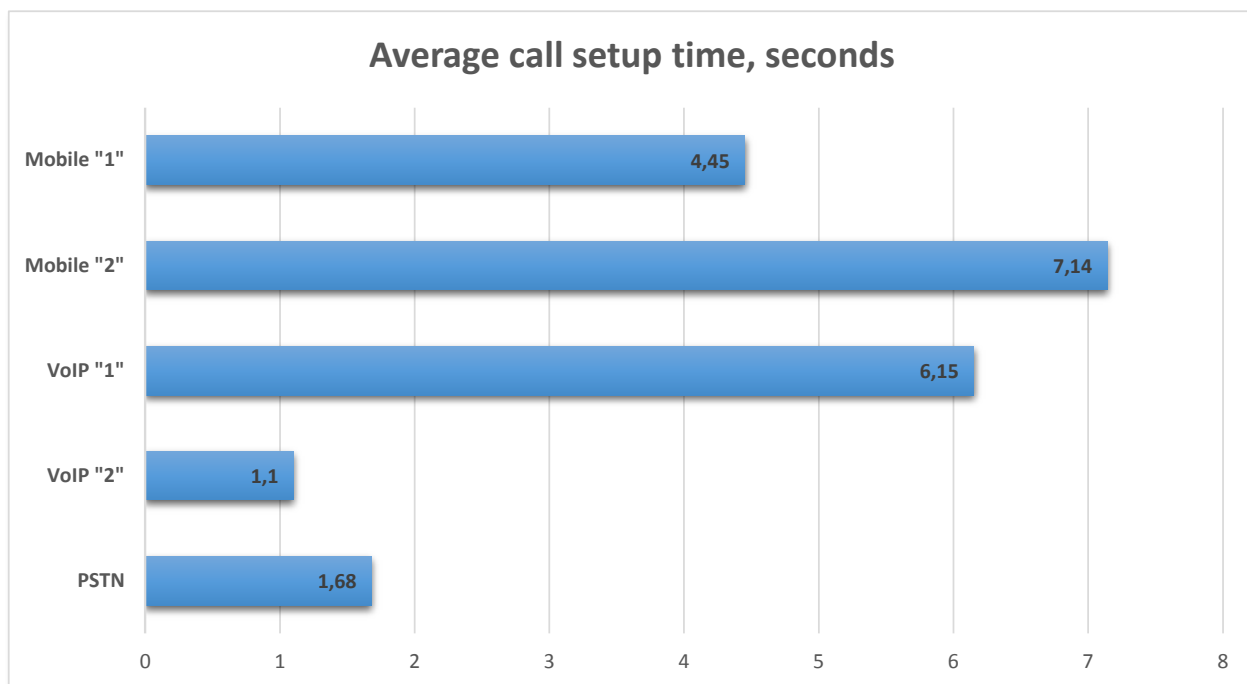|	
Mobile "1"	4,45
Mobile "2"	7,14
VoIP "1"	6,15
VoIP "2"	1,1
PSTN	1,68

Figure 16: Average call setup time in mobile and fixed networks in Latvia

The reason for high call set up times is because some IP-based systems do not recognise the end of the dialling procedure and still are waiting for the next digit (VoIP "1" case in Figure 16 above). In Latvia, the incumbent operator recommends to their customers to use "#" after the last digit (VoIP "2" case in Fig. 16). Normally it helps but some contradictions with the National numbering plan may be introduced.

At the same time the measurements mentioned above do not indicate the voice quality degradation because the QoS level in internet access connections is relatively high.

Determining voice transmission quality in IP-based networks depends on a complex analysis of all relevant parameters inherent to the connection session such as latency, jitter, packet order, packet discards, etc. Tests carried out in 2009 [37] showed that drop-outs in speech signals and the speech delay of VoIP-based services are significantly worse compared to an ISDN-Reference, and that the MOS-values of VoIP-based services are mainly slightly worse compared to ISDN. The big increase of speech delay in VoIP-based networks leads to a need for echo cancellation, which affects the voice transmission quality negatively.

In general customers would not favour a migration process if it might deny them access to, or degrade the quality of, specific services or be less functional for emergency calls.

9.7 DTMF TRANSMISSION

The quality in VoIP networks is typically good enough to transmit the tones used by DTMF, but in a significant number of cases there are problems with short DTMF signals.

On the voice service provider side it is generally argued that the transmission of DTMF is not a voice service but a data service and for data services the IP-based network provides much more sophisticated solutions with more data reliably being transmitted in a shorter period of time.

Because of this and as a pragmatic solution it is usually recommended to replace the old terminal equipment, which utilises DTMF-based data transmission, with new equipment specifically designed for IP-connectivity.

The IETF has published several RFCs since 2000 aimed at providing a solution to the DTMF issue. These are:

- RFC 2833, "RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals" published in May 2000;
- RFC 4733 (same title) which supersedes RFC 2833 published in December 2006;
- RFC 4734 "Definition of Events for Modem, Fax, and Text Telephony Signals", which provided updates to RFC 4733, published in December 2006;
- RFC 5244 "Definition of Events for Channel-Oriented Telephony Signalling", which also provided updates to RFC 4733, published in June 2008.

Specifications for IP-based interconnection between voice service providers usually require RFC 4733 to be supported.

10   CONCLUSIONS

The analysis of the different technical, strategic and regulatory topics discussed in this ECC Report, as well as the results and conclusions of the PT TRIS survey, makes it possible for the following conclusions to be made regarding the migration from PSTN/ISDN to IP-based networks.

- One can conclude that, following consideration of the main drivers for migration, the migration from PSTN/ISDN to IP-based networks is unavoidable;

- The obsolescence of the PSTN/ISDN network equipment, the lack of vendor support, the increasing operational and maintenance costs, and the shortage of skilled maintenance staff are the major drivers for migration to IP based networks;

- The migration to IP based networks is necessary to meet the ongoing demand for capacity and multiple communications services;

- Several migration strategies exist but the phased solution is the most often selected because it allows for a softer approach with the existing PSTN/ISDN network working for a period in parallel with the IP-based network components as they are phased in. The analysis presented in this ECC Report suggests that full PSTN/ISDN migration will ultimately prevail over temporary or combined solutions;

- Network operators planning migration projects should take into account any potential impacts on both their wholesale and retail customers and they should clearly communicate their migration plans and allow sufficient time for customers to prepare for the migration;

- In the case of wholesale customers, the migration plans should take account of investments already made and, where necessary, provide alternative equivalent wholesale products;

- Migrating from PSTN/ISDN to all IP may impact the quality of voice services, the performance of networks providing telephony services and the reliability of non-voice services offered via PSTN/ISDN;

- Interconnecting IP-based networks will pose challenges and interoperability issues will arise that may not have been a problem for the interconnection of PSTN/ISDN networks. The diversity of types of IP interconnection, of signalling protocols and codecs requires a harmonised approach, broad consultation among key stakeholders and may require regulatory intervention;

- Depending on national regulatory frameworks, actions from regulators may be required to minimise the risk of any negative impact on customers.

ANNEX 1: LIST OF REFERENCE

[1]   ITU-T Y.2261 - Global Information Infrastructure, Internet Protocol Aspects and Next-Generation Networks - Service Aspects: Interoperability of Services and Networks in NGN

[2]   ITU-T Recommendation G.100 - Transmission Systems and Media, Digital Systems and Networks: International telephone connections and circuits – General definitions - Vocabulary for performance and quality of service

[3]   ITU-T Recommendation G.704 - Synchronous frame structures used at 1544, 6312, 2048, 8448 and 44 736 kbit/s hierarchical levels

[4]   ITU-T Recommendation Y.2001: Global Information Infrastructure, Internet Protocol Aspects and Next Generation Networks: Next Generation Networks – Frameworks and functional architecture models

[5]   Description of MSAN taken from Wikipedia: https://en.wikipedia.org/wiki/Multi-service_access_node

[6]   Directive 2002/22/EC of the European Parliament and of the Council of 7 March 2002 on Universal Service and Users' Rights relating to Electronic Communications Networks and Services (Universal Service Directive) as amended by Directive 2009/136/EC.

[7]   Directive 2002/19/EC of the European Parliament and of the Council of 7 March 2002 on Access to and Interconnection of Electronic Communications Networks and Associated Facilities (Access Directive) as amended by Directive 2009/140/EC.

[8]   Directive 2009/136/EC of the European Parliament and of the Council of 25 November 2009 amending Directive2002/22/EC on Universal Service and Users' Rights relating to Electronic Communications Networks and Services, Directive 2002/58/EC concerning the Processing of Personal Data and the Protection of Privacy in the Electronic Communications Sector and Regulation (EC) No 2006/2004 on Cooperation between National Authorities responsible for the Enforcement of Consumer Protection Laws.

[9]   ETSI SR 080 005 – Regulatory Requirements and Public Interests in NGN standardisation. Available at: http://www.etsi.org/deliver/etsi_sr/080001_080099/080005/03.01.01_60/sr_080005v030101p.pdf

[10]   ECC Report 195 – Minimum Set of Quality of Service Parameters and Measurement Methods for Retail Internet Access Services. Available at: http://www.erodocdb.dk/doks/filedownload.aspx?fileid=3977&fileurl=http://www.erodocdb.dk/Docs/doc98/official/pdf/ECCREP195.PDF

[11]   ETSI EG 202 057-1 V1.3.1 (2008-07) - Speech Processing, Transmission and Quality Aspects (STQ); User related QoS parameter definitions and measurements; Part 1: General. Available at: http://www.etsi.org/deliver/etsi_eg/202000_202099/20205701/01.03.01_60/eg_20205701v010301p.pdf

[12]   ETSI EG 202 057-2 V1.3.1 (2009-02) "Speech Processing, Transmission and Quality Aspects (STQ); User related QoS parameter definitions and measurements; Part 2: Voice telephony, Group 3 fax, modem data services and SMS". Available at: http://www.etsi.org/deliver/etsi_eg/202000_202099/20205702/01.03.01_60/eg_20205702v010301p.pdf

[13]   ITU-T Recommendation P.862 - Telephone Transmission Quality, Telephone Installations, Local Line Networks Methods for Objective and Subjective Assessment of Quality. Available at: http://www.itu.int/rec/T-REC-P.862-200102-I/en

[14]   ITU-T Recommendation P.863 - Terminals and Subjective and Objective Assessment Methods. Methods for objective and subjective assessment of speech quality. Available at: https://www.itu.int/rec/T-REC-P.863-201409-I/en

[15]   ECC Recommendation (15)03 - Provision of Comparable Information on Retail Internet Access Service Quality. Available at: http://www.erodocdb.dk/doks/filedownload.aspx?fileid=4145&fileurl=http://www.erodocdb.dk/Docs/doc98/official/pdf/REC1503.PDF

[16]   ETSI EG 202 057-4 V1.1.1 (2005-10) - Speech Processing, Transmission and Quality Aspects (STQ); User related QoS parameter definitions and measurements. Part 4: Internet access. Available at: http://www.etsi.org/deliver/etsi_eg/202000_202099/20205704/01.01.01_60/eg_20205704v010101p.pdf

[17]   ITU-T Recommendation Y.1540: Global Information Infrastructure, Internet Protocol Aspects and Next-Generation Networks, Internet of Things and Smart Cities: Internet protocol aspects – Quality of service and network performance. Available at: https://www.itu.int/rec/T-REC-Y.1540-201607-P/en (TIES Account required)

[18]   Directive 2002/21/EC of the European Parliament and of the Council of 7 March 2002 on a common regulatory framework for electronic communications networks and services (Framework Directive).

[19]   Directive 2009/140/EC of the European Parliament and of the Council of 25 November 2009 amending Directives 2002/21/EC on a common regulatory framework for electronic communications networks and services, 2002/19/EC on access to, and interconnection of, electronic communications networks

and associated facilities, and 2002/20/EC on the authorisation of electronic communications networks and services.

[20] Directive 2009/136/EC of the European Parliament and of the Council of 25 November 2009 amending Directive 2002/22/EC on universal service and users' rights relating to electronic communications networks and services, Directive 2002/58/EC concerning the processing of personal data and the protection of privacy in the electronic communications sector and Regulation (EC) No 2006/2004 on cooperation between national authorities responsible for the enforcement of consumer protection laws.

[21] Directive 2002/58/EC of the European Parliament and of the Council of 12 July 2002 concerning the processing of personal data and the protection of privacy in the electronic communications sector (Directive on privacy and electronic communications).

[22] ECC Report 193 - Emergency Calls in VoIP Environment: Compilation of Recent Studies – Available at: http://www.erodocdb.dk/Docs/doc98/official/pdf/ECCREP193.PDF

[23] Convention on the Rights of Persons with Disabilities (CRPD). Available at: https://www.un.org/development/desa/disabilities/convention-on-the-rights-of-persons-with-disabilities.html

[24] IETF RFC 3261 - Session Initiation Protocol. Available at: https://www.rfc-editor.org/rfc/pdfrfc/rfc3261.txt.pdf

[25] IETF RFC 3372 - Session Initiation Protocol for Telephones (SIP-T): Context and Architectures. Available at: https://www.rfc-editor.org/rfc/pdfrfc/rfc3372.txt.pdf

[26] ITU-T Recommendation H.323: Packet-based multimedia communications systems - Audiovisual and Multimedia Systems. Available at: https://www.itu.int/rec/dologin_pub.asp?lang=e&id=T-REC-H.323-200912-I!!PDF-E&type=items

[27] WG NaN Green Paper – Long Term Evolution in Numbering, Naming and Addressing 2012-2022. Available at: http://www.cept.org/files/4591/documents/WG%20NaN%20Green%20Paper%20-%20Long%20Term%20Evolution%20in%20NNA%202012-2022%20-%20Stockholm%20outcome.pdf

[28] RFC 3966 - The tel URI for Telephone Numbers – December 2004 - https://www.ietf.org/rfc/rfc3966.txt

[29] ITU-T Recommendation E.123 - Notation for national and international telephone numbers, e-mail addresses and web addresses - https://www.itu.int/rec/dologin_pub.asp?lang=e&id=T-REC-E.123-200102-I!!PDF-E&type=items

[30] ITU-T Recommendation E.164 - The international public telecommunication numbering plan-https://www.itu.int/rec/dologin_pub.asp?lang=e&id=T-REC-E.164-201011-I!!PDF-E&type=items

[31] RFC 3761 -The E.164 to Uniform Resource Identifiers (URI) Dynamic Delegation Discovery System (DDDS) Application (ENUM) - https://www.ietf.org/rfc/rfc3761.txt

[32] BEREC report "Case Studies on Migration from POTS/ISDN to IP on the Subscriber Access Line in Europe", 2016

[33] Deutsche Telekom Innovation News Article - Another milestone reached in the IP migration. Dec 2015. Available at: https://www.telekom.com/innovation/295928

[34] SIP-PBX / Service Provider Interoperability : SIPconnect 1.1 Technical Recommendation. SIP Forum Document Number: TWG-2. Available at: http://www.sipforum.org/component/option,com_docman/task,doc_download/gid,474/Itemid,75/

[35] SIP Forum – Fax Over IP Task Group: Problem and Recommendation Statement. Available at: http://ftp.tiaonline.org/TR-30/TR-30.1/Public/TR-30.1/2010-04%20Baltimore/T.38%20SIP-SDP%20Recommendation%20Statement%20%282010-05-24%29-clean.pdf

[36] International Interconnection Forum for Services Over IP : Technical Specification for Fax over IP service (Release 3.0, May 2013). Available at: http://i3forum.org/wp-content/uploads/2013/06/i3F-Technical-FoIP-Rel-3-FINAL-2013-4-23.pdf

[37] Reports in the ongoing Benchmark in Connect – Zafaco GmbH – December 20019: http://www.zafaco.de/fileadmin/downloads/2009_12_connect_zafaco_NGN_Test_EN.pdf

[38] Thomas M. Chen, From Circuit Switched to IP-based Networks, Chapter in Encyclopaedia of Multimedia Technology and Networking, 2nd ed., M. Pagani (ed.), Idea Group Publishing

[39] Telecommunications Network and Service Architectures Principles, Concepts and Architectures, EFORT 2010

[40] S. Dornheim, Managing the PSTN Transformation, CRC Press, 2015

[41] Chaesub Lee, Migration scenarios from legacy networks to NGN in developing countries, ITU-T Technical paper, March 2013

[42] Recommendation ITU-T Y.2261 (2006), PSTN/ISDN evolution to NGN

[43] Wikipedia: http://en.wikipedia.org/wiki/Public_switched_telephone_network

[44] Recommendation ITU-T I.324 (1991), ISDN network architecture.

[45] Recommendation ITU-T Q.1290 (1998), Glossary of terms used in the definition of intelligent network.

[46] R. Freeman, Fundamentals of telecommunications, 2005

[47] Wikipedia: https://en.wikipedia.org/wiki/Integrated_Services_Digital_Network

[48] Wikipedia: https://en.wikipedia.org/wiki/Voice_over_IP

[49] J. Hallock, A Brief History of VoIP, November 2004

[50] Cisco, Quality of Service for Voice over IP

[51] Keymile, Migration scenarios of voice on broadband access networks, white paper, 2009

[52] BEREC report "Case studies on IP-based interconnection for voice services in the European Union" , 2015

ANNEX 2: RESULTS OF QUESTIONNAIRES

This Annex is contained in a separate accompanying document entitled – "ECC Report 265 – Annex 2"

ECC Report 265 - Annex 2

Results of Survey: Migration from PSTN/ISDN to
Next Generation Networks (NGN)

Questionnaire prepared by ECC/WG NaN/
Project Team Technical Regulatory Issues (PT TRIS)

Approved 31 May 2017

TABLE OF CONTENTS

INTRODUCTION

In the last few years, incumbent operators in many CEPT countries have announced plans for phasing out PSTN and ISDN services with a view to a full migration to an IP-based NGN. In overall terms these plans have not been fully implemented as fast as was originally anticipated and many ambitious migration plans have either been shelved or redefined.

There are several reasons which may explain why PSTN/ISDN has not been phased out earlier including:

- An unconvincing business case for making a full transition to IP;
- Investment in mobile and broadband given higher priority;
- PSTN/ISDN has proved to be very reliable with few problems;
- Long standardisation process for new systems (IMS/NGN).

In order to develop a better understanding of the issues, this questionnaire, developed by the CEPT/ECC/WG NaN Project Team on Technical Regulatory Issues (PT TRIS), aimed to gather information from CEPT administrations on migration strategies for PTSN/ISDN to all-IP networks in a European context including on the associated challenges and regulatory issues.

There were initially only 11 responses to the Questionnaire and therefore it was decided at the last meeting of the ECC Working Group on Numbering and Networks (WG NaN) in Oslo (October 2014) to extend the deadline to 31 October 2014. An additional 8 responses were received making a total of 19.

QUESTION 1

Does your incumbent fixed network operator have an official plan for PSTN migration? (e.g. by making a public announcement or releasing information to the media?)

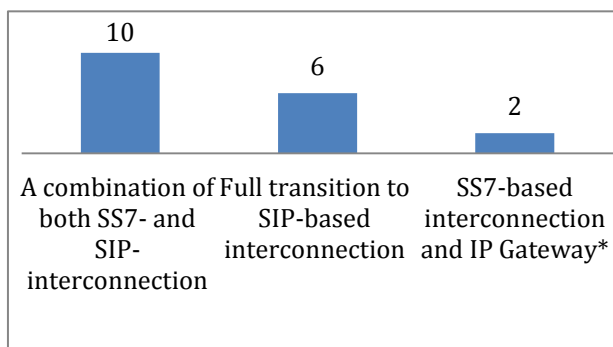| Country	Response	Comments
Switzerland	Yes	
Lithuania	No	
Russian Federation	Yes	
Slovenia	No	
Austria	Yes	
Bulgaria	Yes	
Malta	No	
Romania	Yes	
Portugal	No	
Czech Republic	No	
Norway	Yes	
Germany	Yes	
Cyprus	No	
Croatia	Yes	
Montenegro	Yes	
Latvia	No	
Denmark	Yes	
France	No	
Ireland	No	NGA is an overlay and PSTN is being retained.
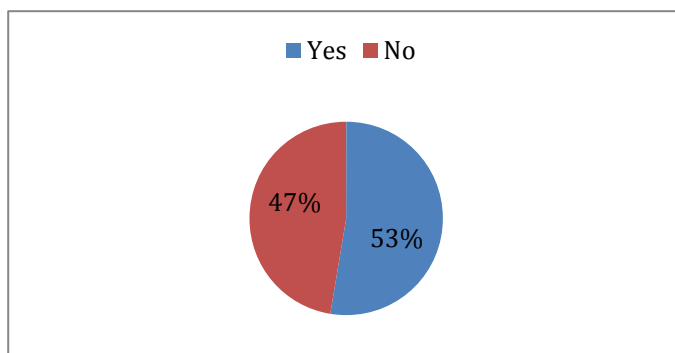
QUESTION 1.1

If yes to Question 1, what is the time schedule for expected shut-down of PSTN-services?

Country	Response	Sub Question: If yes to Question 1.1, are there major differences between the incumbent and other PSTN-providers? Please specify
Switzerland	End of 2017	No major differences, just different time schedules
Russian Federation	The time schedule for shut-down is not defined yet	incumbent provider has more finance
Austria	No official shut-down date been announced. We expect PSTN services to be around for some further time as the incumbent NGN has been introduced with feature parity, i.e. the majority of PSTN services will also be available with the IP-based NGN.	No information available.
Bulgaria	There is no schedule for a shutdown of PSTN services. The incumbent has announced a forecast that more than 1 million end users will be able to receive more services based on broadband access until the end of 2014. First the 3 biggest towns will be covered and along the years all other areas.	No
Romania	Yes, Telekom Romania (PSTN provider in Romania) announced the deadline for shutting-down of PSTN services on 31. Dec. 2017	In Romania, there is no other PSTN provider
Norway	2017, but is likely to be extended	No
Germany	Until 2018	The plans deviate significantly. Some providers do not yet plan to migrate.
Croatia	End of 2015	Alternative operators are already on IP
Montenegro	The end of the 2015.	There are no more PSTN providers. Only Crnogorski Telekom provides it.
Denmark	No public announcement, but it has been communicated that PSTN is expected to be phased out over the years up to 2020	No information

QUESTION 2

How will future interconnection between telephony providers be supported in your country?

Country	Response
Switzerland	A combination of both SS7- and SIP-interconnection
Lithuania	SS7-based interconnection will continue (via gateway)
Russian Federation	A combination of both SS7- and SIP-interconnection
Slovenia	Full transition to SIP-based interconnection
Austria	Full transition to SIP-based interconnection
Bulgaria	A combination of both SS7- and SIP-interconnection
Malta	
Romania	A combination of both SS7- and SIP-interconnection
Portugal	A combination of both SS7- and SIP-interconnection
Czech Republic	A combination of both SS7- and SIP-interconnection
Norway	A combination of both SS7- and SIP-interconnection
Germany	A combination of both SS7- and SIP-interconnection
Cyprus	A combination of both SS7- and SIP-interconnection
Croatia	Full transition to SIP-based interconnection
Montenegro	Full transition to SIP-based interconnection
Latvia	Full transition to SIP-based interconnection
Denmark	A combination of both SS7- and SIP-interconnection
France	Full transition to SIP-based interconnection
Ireland	Currently only SS7-based interconnection and IP Gateway. Likely to be a combination of both SS7- and SIP-interconnection in future.

* Likely to be a combination of both SS7 and SIP-interconnection in Ireland in the future.

QUESTION 3

In relation to your response to Question 2 above, please specify which alternative(s) are mandated by the NRA?

| Country	Response	Sub-Question: Do you foresee major technical problems or threats in adopting SIP/IP-based interconnection? If so, please explain.
Switzerland		Yes, managing SIP credentials
Lithuania	SS7-based interconnection via gateway is short term solution. NRA is not defined now any alternatives but in future it could define.	No
Russian Federation	At present, the NRA does not mandate any alternatives. Providers solve these issues by themselves.	We see the problems related to safety, the QoS and overload
Slovenia		No
Austria	In fact, the migration from PSTN to an IP based NGN of the incumbent fixed network operator A1 Telekom Austria (A1TA) was finished in 2013. TKK's (Telekom-Control-Kommission) decision on M3/2007 on interconnection is technology neutral, but imposes interconnection at a single PoI (point of interconnection) with one regulated rate calculated based on NGN. Hence the interconnection can be requested based on TDM or IP. At the moment it is not foreseeable, how long the transition phase, where both SS7 and IP-interconnections are deployed, will last.	Currently some mobile and fixed alternative network operators are already interconnecting via IP for e.g. voice services. Beside SIP some mobile operators are also using BICC for this purpose. Furthermore SIGTRAN is widely deployed for the transport of SS7 over IP.
Bulgaria	SS7 interconnection is mandatory. SIP interconnection is under free trade agreements. Currently CRC is working on implementing rules for IP interconnection.	Currently CRC and providers carry out consultation meetings as the main goals are to define: -The appropriate SIP protocol version applicable to all providers. -Quality parameters for interconnection and services.
Malta		
Romania	There is not a NRA decision, it is operator's option depending on market demand.	Based on discussions with the operator, we do not foresee major technical problems. However, there is no pressure on switching to SIP/IP-interconnection either.
Portugal	It is foreseen to mandate the provision of both solutions (SS7 and SIP trunk) to the SMP (incumbent operator).	The technical solution should be agreed between operators. The NRA will act as mediator.
Czech Republic	Both	No

Country	Response	Sub-Question: Do you foresee major technical problems or threats in adopting SIP/IP-based interconnection? If so, please explain.
Norway	No specific technical interface is mandated, but interconnection must be offered.	No severe problems or threats are experienced so far, but many options and variants of SIP/IP-based interconnection can be a complicating factor.
Germany	SIP based interconnection. Detail specifications have been worked out by the national interconnection standardization group AKNN. (See http://www.aknn.de/index.php/1731/0/).	No
Cyprus	N/A	N/A
Croatia	It is foreseen to mandate the provision of both solutions (SS7 and SIP trunk) to the SMP (incumbent operator).	The technical solution should be agreed between operators. The NRA will act as mediator
Montenegro		Crnogorski Telekom will use experience of other Natco's and operators for applied solution. As it was happened, fraud can be easier made in this environment.
Latvia	There are not alternatives mandated by the NRA, now. NRA is planning to mandate a combination of both SS7- and SIP-interconnection for migration from PSTN to NGN in transitional stage, but full transition to SIP based interconnection in future.	NRA does not foresee major technical problems or threats in adoption SIP/IP-based interconnection, jet.
Denmark	The incumbent having both PSTN- and VoIP-end users is mandated to give access to both SS7- and SIP-interconnect.	No
France	N/A	The main issue with the transition from SS7 to SIP at the interconnection level is to make sure all services traditionally provided over SS7 (especially for business customers like signalling and latency-sensitive services) are implemented properly in IP to ensure a continuity of service to end-users.
Ireland	No specific technical interface is mandated, but interconnection must be offered.	

QUESTION 4

What are the main drivers for the PSTN-migration? (tick more than one box if appropriate)

Country	Response
Switzerland	Old equipment Vendor support Staff competence
Lithuania	Operational costs Old equipment Vendor support
Russian Federation	Old equipment Vendor support Staff competence Other: new services, attraction of investments
Slovenia	Operational costs Old equipment Vendor support
Austria	Operational costs Old equipment Vendor support This question should be addressed towards operators, so the answer is best guess from a regulator's point of view.
Bulgaria	Operational costs
Malta	
Romania	Operational costs Old equipment Vendor support
Portugal	Operational costs Vendor support Staff competence
Czech Republic	Other: No information yet
Norway	Operational costs Vendor support
Germany	Operational costs Old equipment Vendor support Staff competence
Cyprus	Operational costs Old equipment Vendor support

Country	Response
Croatia	Operational costs Old equipment Vendor support
Montenegro	Operational costs Old equipment Vendor support
Latvia	Operational costs Old equipment Vendor support
Denmark	Operational costs Old equipment Vendor support Staff competence
France	Operational costs Old equipment Vendor support Staff competence Other: At the interconnection level, the migration towards IP is driven by the fact that SS7 interconnection, for historical reasons, requires much more interconnection points than IP-based interconnections.
Ireland	Currently N/A. Ultimately operational costs and lack of vendor support may force migration.
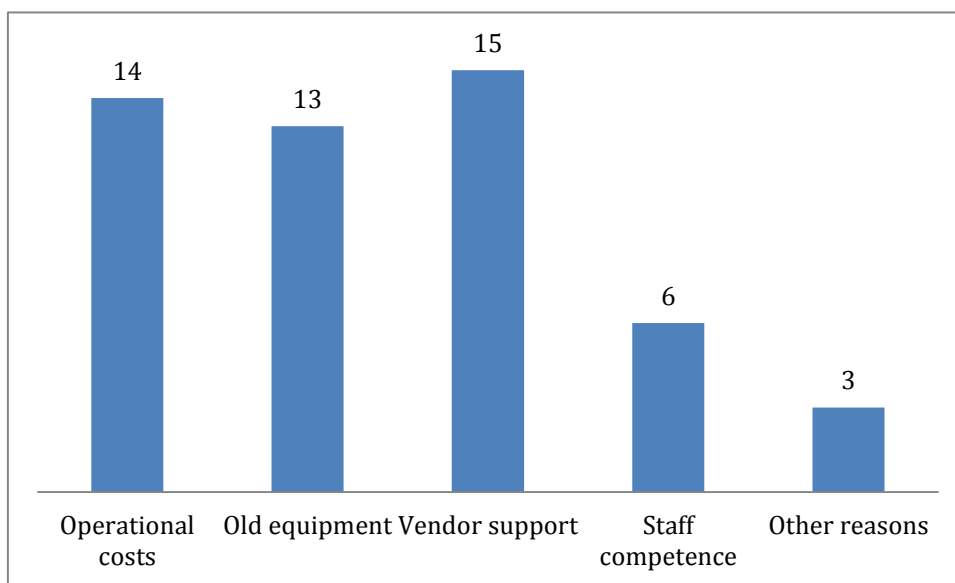
QUESTION 5

What kind of migration strategy is planned?

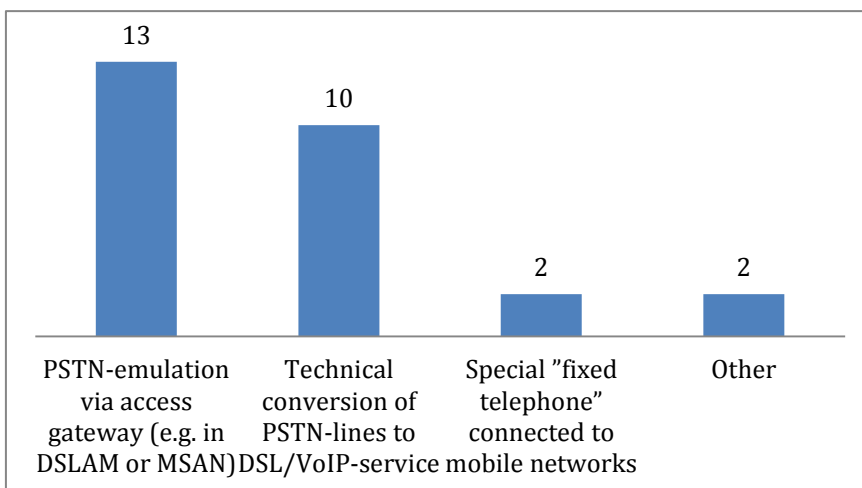| Country	Response
Switzerland	
Lithuania	A combination of technical and commercial migration (see also Questions 6 and 7).
Russian Federation	A combination of technical and commercial migration (see also Questions 6 and 7).
Slovenia	A combination of technical and commercial migration (see also Questions 6 and 7).
Austria	Technical migration i.e. the end customer will keep the existing fixed voice service with no or only minor changes (see also Question 6).
Bulgaria	Technical migration i.e. the end customer will keep the existing fixed voice service with no or only minor changes (see also Question 6).
Malta	
Romania	Technical migration i.e. the end customer will keep the existing fixed voice service with no or only minor changes (see also Question 6).
Portugal	Commercial migration i.e. the end customer must change his subscription to alternative voice services (see also Question 7).
Czech Republic	A combination of technical and commercial migration (see also Questions 6 and 7).
Norway	A combination of technical and commercial migration (see also Questions 6 and 7).
Germany	A combination of technical and commercial migration (see also Questions 6 and 7).
Cyprus	A combination of technical and commercial migration (see also Questions 6 and 7).
Croatia	A combination of technical and commercial migration (see also Questions 6 and 7).
Montenegro	A combination of technical and commercial migration (see also Questions 6 and 7).
Latvia	A combination of technical and commercial migration (see also Questions 6 and 7).
Denmark	A combination of technical and commercial migration (see also Questions 6 and 7).
France	
Ireland	N/A
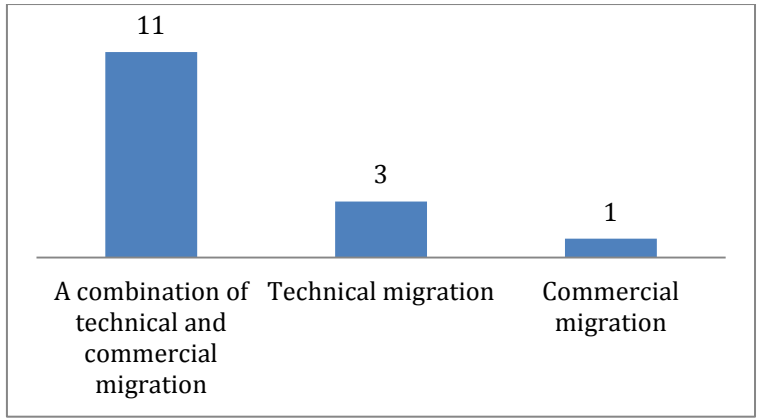
QUESTION 6

If technical migration is planned, what type of solution? (tick more than one box if appropriate)

Country	Response
Switzerland	Technical conversion of PSTN-lines to DSL/VoIP-service?
Lithuania	PSTN-emulation via access gateway (e.g. in DSLAM or MSAN)
Russian Federation	Technical conversion of PSTN-lines to DSL/VoIP-service technical conversion of PSTN-lines to DSL/VoIP-service
Slovenia	PSTN-emulation via access gateway (e.g. in DSLAM or MSAN) Technical conversion of PSTN-lines to DSL/VoIP-service
Austria	PSTN-emulation via access gateway (e.g. in DSLAM or MSAN)
Bulgaria	PSTN-emulation via access gateway (e.g. in DSLAM or MSAN) Technical conversion of PSTN-lines to DSL/VoIP-service Special "fixed telephone" connected to mobile networks
Malta	
Romania	PSTN-emulation via access gateway (e.g. in DSLAM or MSAN) Technical conversion of PSTN-lines to DSL/VoIP-service
Portugal	
Czech Republic	PSTN-emulation via access gateway (e.g. in DSLAM or MSAN) Technical conversion of PSTN-lines to DSL/VoIP-service
Norway	PSTN-emulation via access gateway (e.g. in DSLAM or MSAN) Special "fixed telephone" connected to mobile networks? Comment: PSTN-emulation not yet decided, but is likely in some areas.
Germany	PSTN-emulation via access gateway (e.g. in DSLAM or MSAN)
Cyprus	Technical conversion of PSTN-lines to DSL/VoIP-service
Croatia	PSTN-emulation via access gateway (e.g. in DSLAM or MSAN) Technical conversion of PSTN-lines to DSL/VoIP-service Comment: Technical conversion of PSTN-lines to DSL/VoIP-service is Main migration scenario, but PSTN-emulation via access gateway is used in some cases for voice only customers.
Montenegro	PSTN-emulation via access gateway (e.g. in DSLAM or MSAN) Technical conversion of PSTN-lines to DSL/VoIP-service Comment: Technical conversion of PSTN-lines to DSL/VoIP-service is Main migration scenario, but PSTN-emulation via access gateway in some cases for voice only customers.
Latvia	PSTN-emulation via access gateway (e.g. in DSLAM or MSAN) Technical conversion of PSTN-lines to DSL/VoIP-service

| Country	Response
	Other: VoIP via fibre (GPON)
Denmark	PSTN-emulation via access gateway (e.g. in DSLAM or MSAN) Other: Technical migration for POTS services using PSTN-simulation
France	At the access level, no migration is currently planned.
Ireland	N/A

QUESTION 7

If commercial migration is planned, what type of alternatives will be offered to existing PSTN-subscribers?
(tick more than one box if appropriate)

Country	Response
Switzerland	Alternative voice service via broadband access (e.g. VoIP-services over xDSL, cable-TV, fibre) Comment: IP telephony over DSL or fibre
Lithuania	Alternative voice service via broadband access (e.g. VoIP-services over xDSL, cable-TV, fibre)
Russian Federation	Alternative voice service via broadband access (e.g. VoIP-services over xDSL, cable-TV, fibre)
Slovenia	Alternative voice service via broadband access (e.g. VoIP-services over xDSL, cable-TV, fibre)
Austria	
Bulgaria	
Malta	
Romania	N.A.
Portugal	Alternative voice service via broadband access (e.g. VoIP-services over xDSL, cable-TV, fibre) Alternative mobile services Other: services for customers without broadband access and outside mobile coverage (e.g. fixed radio access, satellite).
Czech Republic	Alternative voice service via broadband access (e.g. VoIP-services over xDSL, cable-TV, fibre) Alternative mobile services
Norway	Alternative voice service via broadband access (e.g. VoIP-services over xDSL, cable-TV, fibre) Alternative mobile services
Germany	Alternative voice service via broadband access (e.g. VoIP-services over xDSL, cable-TV, fibre) Alternative mobile services
Cyprus	Alternative voice service via broadband access (e.g. VoIP-services over xDSL, cable-TV, fibre)
Croatia	Alternative voice service via broadband access (e.g. VoIP-services over xDSL, cable-TV, fibre)
Montenegro	Alternative voice service via broadband access (e.g. VoIP-services over xDSL, cable-TV, fibre) Alternative mobile services Comment: VoBB is main scenario. Fibre rollout is in progress in Crnogorski Telekom, but alternative mobile services not initially planned, only in limited situation if it is not possible to

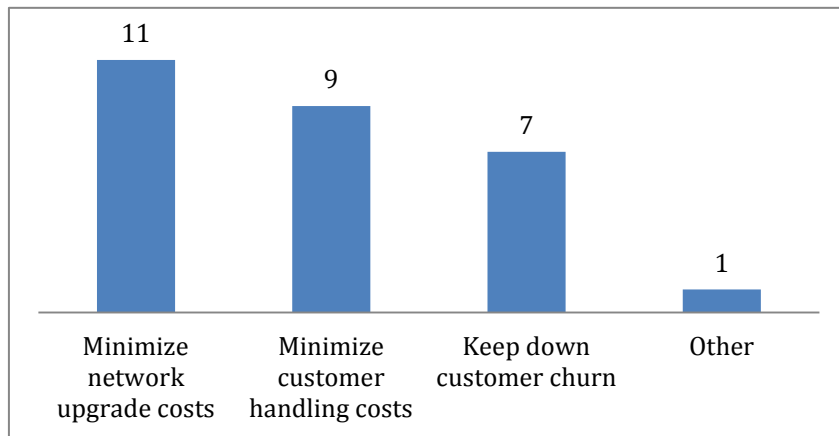| Country	Response
	provide voice service based on regular migration scenarios.
Latvia	Alternative voice service via broadband access (e.g. VoIP-services over xDSL, cable-TV, fibre)
Denmark	Alternative voice service via broadband access (e.g. VoIP-services over xDSL, cable-TV, fibre) Comment: POTS over broadband will be a commercial alternative to POTS + broadband. ISDN alternatives will include SIP-trunk + IP-centrex
France	N/A
Ireland	N/A
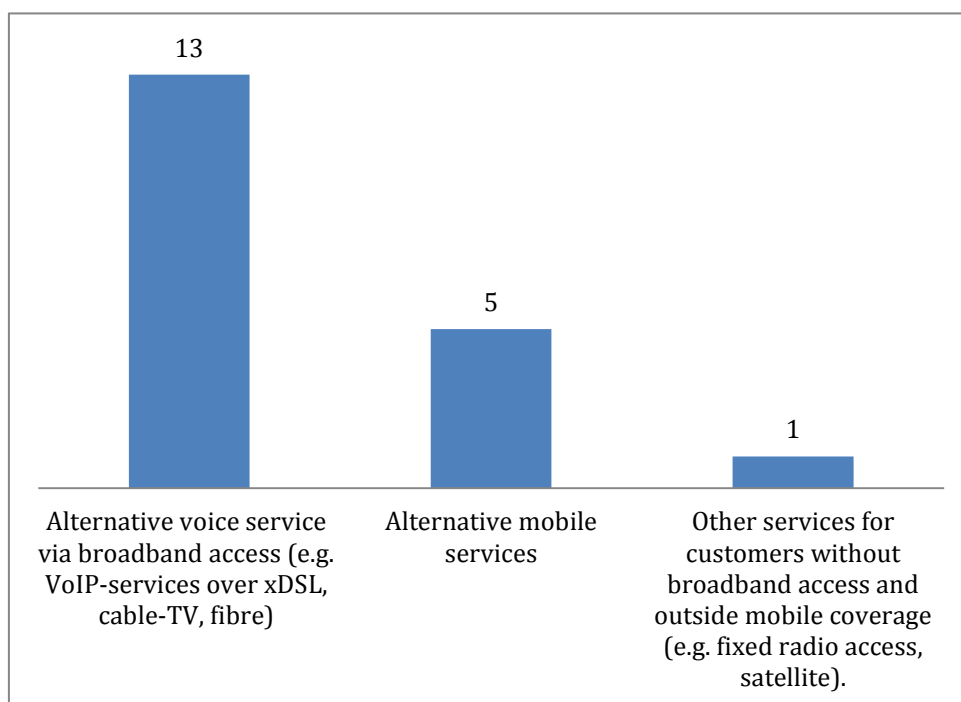
QUESTION 8

What are the main reasons for choosing a technical and/or commercial migration strategy? (tick maximum two alternatives)

| Country	Comment
Switzerland	Other: Due to the general technological evolution the operators are finally forced to make technological migrations
Lithuania	Minimize network upgrade costs Minimize customer handling costs
Russian Federation	Minimize network upgrade costs Minimize customer handling costs Keep down customer churn
Slovenia	Minimize network upgrade costs
Austria	Minimize customer handling costs Keep down customer churn Comment: Again, this question should be addressed towards operators, so the answer is best guess from a regulator's point of view.
Bulgaria	no information
Malta	
Romania	Minimize customer handling costs Keep down customer churn
Portugal	Minimize network upgrade costs Keep down customer churn
Czech Republic	Minimize network upgrade costs
Norway	Minimize network upgrade costs Keep down customer churn
Germany	Keep down customer churn
Cyprus	Minimize network upgrade costs Minimize customer handling costs Minimize marketing costs
Croatia	Minimize network upgrade costs Minimize customer handling costs
Montenegro	Minimize network upgrade costs Minimize customer handling costs
Latvia	Minimize network upgrade costs Minimize customer handling costs
Denmark	Minimize network upgrade costs Minimize customer handling costs

Country	Comment
	Keep down customer churn
France	
Ireland	N/A

QUESTION 9

Do operator(s) in your country have a universal service obligation (USO) to provide public telephone service?

Country	Response
Switzerland	Yes
Lithuania	Yes
Russian Federation	Yes
Slovenia	Yes
Austria	Yes
Bulgaria	Yes
Malta	
Romania	No
Portugal	Yes
Czech Republic	No
Norway	Yes
Germany	Yes
Cyprus	Yes
Croatia	Yes
Montenegro	Yes
Latvia	Yes
Denmark	Yes
France	Yes
Ireland	Yes
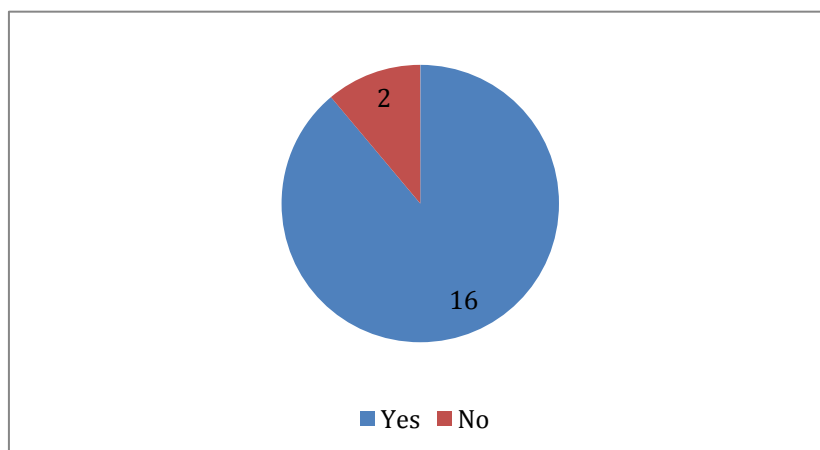
QUESTION 9.1

If yes, how is this service obligation required to be delivered?

Country	Response
Switzerland	Only by PSTN + other technologies in special cases
Lithuania	Only by PSTN
Russian Federation	Only by PSTN
Slovenia	By all technologies incl. VoIP (technology neutral)
Austria	Only by PSTN + other technologies in special cases
Bulgaria	By all technologies incl. VoIP (technology neutral)
Portugal	By all technologies incl. VoIP (technology neutral)
Norway	Only by PSTN + other technologies in special cases
Germany	By all technologies incl. VoIP (technology neutral)
Cyprus	Only by PSTN + other technologies in special cases
Croatia	By all technologies incl. VoIP (technology neutral)
Montenegro	By all technologies incl. VoIP (technology neutral)
Latvia	Only by PSTN + other technologies in special cases
Denmark	By all technologies incl. VoIP (technology neutral)
France	By all technologies incl. VoIP? (technology neutral)
Ireland	Only by PSTN + other technologies in special cases (e.g. fixed mobile solution in very rural areas)
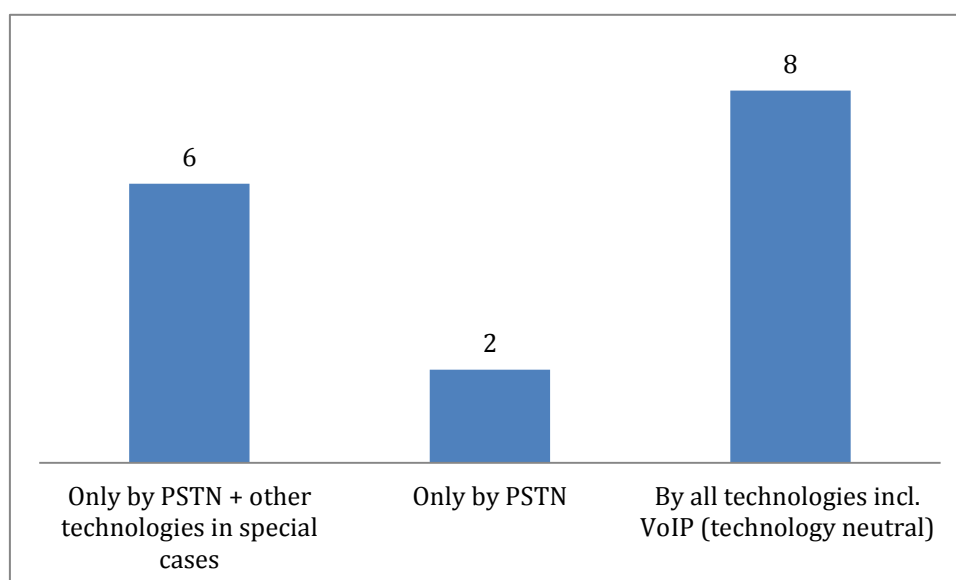
QUESTION 10

Also, if you answered "Yes" to Question 9 does the USO include technical QoS requirements? Please provide further information below.

Country	Response
Switzerland	The SU provider must provide yearly to the NRA a QoS report containing: - Network installation and repair time statistics - Availability of network access statistics - Telephony and narrow band services quality statistics - Wide band services (internet access) quality statistics - Billing and helpdesk performance statistics - Public payphones quality statistics
Lithuania	Yes
Russian Federation	I cannot say anything on this subject
Slovenia	QoS parameters for USO services are mandated by General Act.
Austria	Please refer to https://www.rtr.at/en/tk/UDVerordnung.
Bulgaria	QoS requirements according to USD 2002/22/EO
Malta	
Romania	
Portugal	Yes. Please consider Annexes 2 in sections 2.2 (Telephone service at a fixed location), 3.2 (Public payphones) and 4.2 (Telephone directory enquiry service) contained in the following link: http://www.anacom.pt/streaming/final_decision_USP_07february2012.pdf?contentId=1122782&field=ATTACHED_FILE
Czech Republic	
Norway	No specific technical QoS-requirements apart from providing access to a "digital network"
Germany	No.
Cyprus	YES. There are in place QoS indicators about voice telephony. (e.g. Time for initial line installation, fault rate, Fault recovery time, call failure rate, service response time, call recovery time, allegations of bill errors).
Croatia	QoS parameters for USO services are mandated by Ordinance on universal services. (http://www.hakom.hr/UserDocsImages/2011/propisi_pravilnici_zakoni/Pravilnik%20o%20univerzalnim%20uslugama%20u%20elektroničkim%20komunikacijama%20NN%2023_09.PDF)
Montenegro	Yes, it is prescribed to fulfil the requirements in accordance by Law and the Rule on Quality of Universal Services In accordance to the Law on electronic communications, the services of the Universal Service shall include: 1) meeting any reasonable user's request for the access to public electronic communications network and publicly available electronic communications services on a

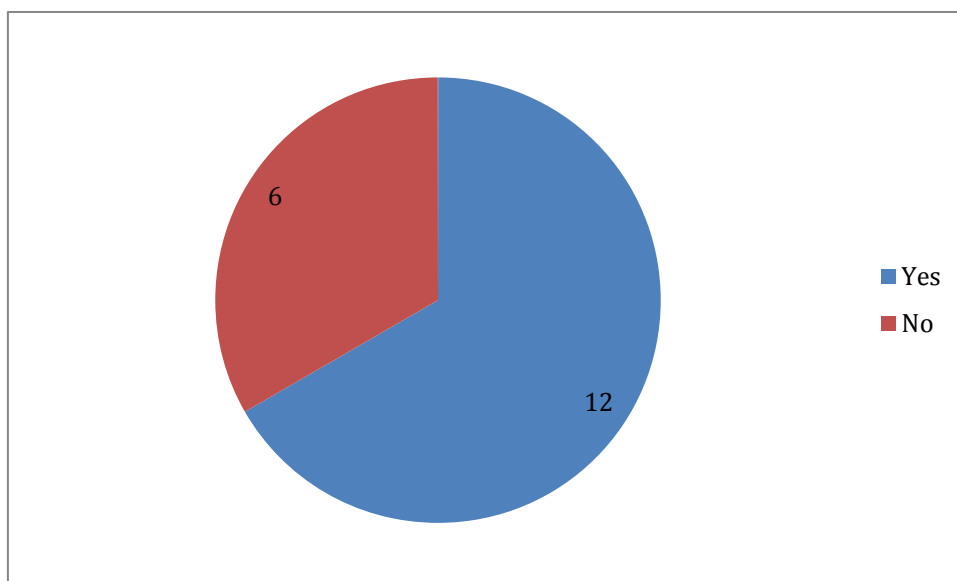| Country	Response
	fixed location, which enables the use of voice communication and data transfer rate which enables functional Internet access; 2) providing the service of universal telephone directory (hereinafter: "Universal Directory") and universal enquiry service for subscriber numbers (hereinafter: „Universal Enquiry Service"); 3) use of public pay telephones or other publicly available points for provision of voice services at any time, in accordance with the reasonable requests and needs of end users, in terms of geographic coverage and service quality; 4) special measures and benefits for the persons with reduced mobility and persons with disabilities, including the access to emergency services, enquiry service for subscriber numbers and directory of subscribers, which enables them to have the same possibilities of access to the publicly available telephone services as the other end users as well as the appropriate selection of operators available to the majority of end users. However, it is prescribe by specialized regulation of Government of Montenegro to exclude the obligation of providing public phones within USO (point 3). Also, Telenor (which is obliged to provide USO 1. and 4. points above) is mobile operator which currently does its USO obligation via FWA (Fixed Wireless Access). The other operator has been elected to provide USO obligations from points 2. i 4.
Latvia	NRA has defined the following parameters for the universal service: - Supply time for initial connection - Unsuccessful call ratio; - Fault repair time - Call set up time; - Response times for operator services; - Response times for directory enquiry services; Values of universal service quality parameters are determined by Regulator, except the parameter "Supply time for initial connection", that is declared by universal service provider for the current year.
Denmark	No
France	Yes, the universal service specifications include QoS requirements, like time limits for connection of new customers and service delivery, time limits for service reestablishment after a breakdown, and more technical requirements like the call set up time and the call failure rate.
Ireland	Yes. Refer to ComReg Document 14/71 on www.comreg.ie

QUESTION 11

Are PSTN-Services being sold without restrictions to new customers in your country?

Country	Response
Switzerland	No
Lithuania	Yes
Russian Federation	Yes
Slovenia	No
Austria	Yes
Bulgaria	Yes
Malta	
Romania	Yes
Portugal	Yes
Czech Republic	Yes
Norway	Yes
Germany	No
Cyprus	Yes
Croatia	No
Montenegro	No
Latvia	Yes
Denmark	No
France	Yes
Ireland	Yes

QUESTION 11.1

Are PSTN-services being sold without restrictions to new customers in your country?

Country	Response
Switzerland	No
Lithuania	Yes
Russian Federation	Yes
Slovenia	No
Austria	Yes
Bulgaria	Yes
Malta	
Romania	Yes
Portugal	Yes
Czech Republic	Yes
Norway	Yes
Germany	No
Cyprus	Yes
Croatia	No
Montenegro	No
Latvia	Yes
Denmark	No
France	Yes
Ireland	Yes

QUESTION 11.2

If no to Question 11.1, what restrictions apply? (tick more than one box if appropriate)

Country	Response
Switzerland	Only available in specific geographic areas
Slovenia	Only available in specific geographic areas Comment: In other areas Telco already made migration (replace legacy telephone exchange with Softswitch). Incumbent operator has not officially announced PSTN/ISDN migration, but unofficially migration is expected in 2018.
Germany	Other restrictions Please specify in the "Remarks" field below: PSTN is only offered, when technically and commercially feasible.
Croatia	Only available in specific geographic areas Other: PSTN is only offered, when technically feasible.
Montenegro	Only available in specific geographic areas Comment: PSTN services are available only in specific areas where the migration is not implemented.
Denmark	Other: PSTN-services are sold without restrictions, if available