White Paper

Migration scenarios of voice on broadband access networks

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1 Abstract

This paper presents an overview of the available and widespread public switched telephone network and also at the emerging next generation network (NGN) under the aspect of telephony services in the wireline access network. It examines possible migration scenarios from the TDM-based PSTN to the packet-based NGN.

The terms VoBB, NGN and IMS are clarified in separate sections.

2 Introduction

2.1 Voice over IP (VoIP) Considerations

2.1.1 VoIP implementation approaches

Legacy TDM-based PSTN voice networks evolve to packet-based networks, where voice over IP (VoIP) can be offered by two different approaches:

- Next Generation Network (NGN):
  NGN as a service application transports voice over managed and secured IP networks with well defined guarantees for end user reachability, communication quality, reliability and connectivity, while supporting services inherited from the PSTN.
- Voice over Internet:
  Internet telephony with voice traffic routed on a "best effort" basis as an end-user or peer-to-peer application on top of the Internet, providing only limited service quality.

From an incumbent telephone service provider's perspective, as they adopt the migration to an end-to-end IP-based infrastructure, the NGN approach becomes a more efficient way of routing and delivering voice traffic, as Internet calls share bandwidth on the network with other applications. The focus of this paper is the NGN approach.

2.1.2 Quality of service and VoIP

Looking at quality of service (QoS) in the VoIP application, the core IP network and the access network have to be distinguished.

Concerning the core network, IP QoS is based either on MPLS or can be accomplished by over-provisioning of bandwidth.

Guaranteeing QoS in the access network requires more attention as the available bandwidth is in general much lower than in the core network. The traffic requirements for the voice service are quite predictable, enabling service providers to introduce QoS by static provisioning in the network.

Voice over Internet using SIP allows a clear differentiation between services, which normally will result in a good quality, but quality remains unpredictable and varies over time even during a call.

On the other hand only a call with predictable behaviour can have the guaranteed QoS. This characteristic will be the main differentiation between "voice over Internet" and "voice over NGN".

2.2 Evolution towards the Next Generation Network (NGN)

2.2.1 Technical background

Looking back 10 years ago the only telephony service a residential customer had available was voice service. With an analogue wire line or with ISDN basic access the connection to the telephony network was restricted to one or two narrow band voice channels.

With the advent of the Internet the demand for data transport increased. In the first years, Internet access was restricted to the narrow band voice channel transport. The drawback of this solution for the customer was twofold: First the bandwidth for data transport was very limited, and second the usage of Internet was only possible as an alternative to telephony usage, at least for analogue wire line access.

Bringing DSL to the customers solved this twofold problem with increasing bandwidth and now concurrent use of telephony and Internet services.

On the other hand Internet service provisioning implied for the service providers the build-up of a data network based on packet transport in parallel to the voice network based on TDM transport.

The ever increasing data bandwidth of broadband access lines brings now the potential of new services and also the possibility of voice over IP (VoIP) transport.

2.2.2 Commercial background

Traditional network and service providers are nowadays confronted with the risk of loosing their revenue by the following factors:

- Competition from new market entrants and decreasing customer loyalty
- Increasing replacement of the fixed wire line access by mobile access
- Substitution of revenue generating TDM services such as leased lines and telephony access by packet based services (e.g. VoIP)

To remain competitive the network and service providers are forced to take the following measures:

- CAPEX reduction by converging the TDM-based telephony and packet-based Internet networks, i.e. by sharing one common network infrastructure
- OPEX reduction by simplified operation and maintenance
- Increasing the revenue by offering new (multimedia) services

In other words the two existing parallel networks have to be converged to a single and universal network offering all services a customer needs now and in the future. This network is, from today's point of view, the next generation network (NGN).

The ITU-T organisation has taken the effort to provide a recommendation framework covering the next generation network in the Y.2000 series.

2.2.3 NGN requirements

What conditions does the NGN need to fulfil to provide the customers their requested services and to give the network and service providers the competitive advantage over their competition?

- Continuity of the existing carrier-grade services offered to customers with the same quality, security and reliability
- Interworking and interoperability between existing and new networks and systems
- Flexibility to incorporate new services
- Quality of service (QoS) to guarantee the service level agreements (SLA) for different traffic conditions and services
- Service continuity in the presence of faults within the network (survivability), offering of life-lines (e. g. for emergency phones)
- Generalised mobility for consistent and ubiquitous provision of services

2.2.4 Migration principles and procedures

A successful migration towards the NGN has to follow some principles:

- The migration of the legacy telephony network to the NGN has to preserve the existing investments as much as possible, e. g. reuse of the copper wires for DSL access, no replacement of the customer telephone sets and PABX's (at least in an initial phase).
- Cost control of the migration process. Bringing broadband data access to the customers also means bringing access network elements closer to the customers. A big amount of network provisioning cost lies in the network access area. Rolling out broadband access should therefore be justified by appropriate services with corresponding revenues. This implicates a stepwise migration procedure.
- The migration steps depend on the specific service provider and market situation, e.g. developing vs. developed country, or actual state of the deployed copper network. The migration steps presented in this white paper are not all mandatory and also have not to be followed in the proposed order.

The migration procedure can be structured in three main steps:

- Optimization and preparation

  Optimize the existing telephone network by a reduction of the number of local and trunk exchanges. Newly deployed access network elements should be prepared for the NGN, e. g. Multi-Service Access Nodes (MSAN) which can realise both, TDM access and packet access.
- Capacity increase

  Deploy NGN network elements to expand the telephone network. NGN and PSTN networks coexist side by side.
- PSTN replacement

  Replace existing PSTN network elements with their equivalent NGN network elements. Customer equipment is replaced by packet based and eventually multimedia service capable equipment. The PSTN will cease to exist.

  Note that network and service providers are not able to control the replacement pace of legacy telephones. Many customers requiring telephone service only will not be willing to replace their familiar handset. Services handling this equipment must be maintained by implementing media gateways in the access network or at the subscriber's premises.

2.3 Explanation of Terms

2.3.1 What is VoBB?

VoBB stands for voice over broadband. VoBB refers to VoIP services that allow end users to make and receive calls over a broadband connection, e. g. a DSL or cable TV connection. End users have direct access to the IP network either with IP capable terminals or with terminal adapters.

VoBB can be used as voice over Internet or as voice over NGN.

2.3.2 Voice over Internet

Voice over Internet is Internet telephony using VoIP routed on a "best effort" basis providing only limited quality and guarantees. With increasing competition, quality of service becomes an issue. Network congestion and outages do occur and affect VoIP quality and reliability. When downloading a large file and making a simultaneous VoIP phone call, interferences may be heard or the call even may be dropped.

Voice over Internet also lacks regulatory and legal requirements such as emergency calls or lawful interception.

2.3.3 Voice over NGN

What is NGN?

ITU-T Y.2001 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. It enables unfettered access for users to networks and to competing service providers and/or services of their choice. It supports generalized mobility which will allow consistent and ubiquitous provision of services to users.

One of the main characteristics of NGN is the decoupling of services and transport, allowing them to be offered separately and to evolve independently. NGN allows the provisioning of both existing and new services independently of the network and the access type used.

In NGN the functional entities may be distributed over the infrastructure. When they are physically distributed, they communicate over open interfaces. Standardised protocols provide the communication between functional entities. Interworking between NGNs of different service providers and between NGN and existing networks such as PSTN, ISDN and GSM is provided by means of gateways.

NGN supports both existing and "NGN aware" end terminal devices. Hence terminals connected to NGN include e.g.

- analogue telephone sets
- fax machines
- ISDN sets
- cellular mobile phones
- GPRS terminal devices
- SIP terminals
- Ethernet phones through PCs
- digital set top boxes
- cable modems

NGN supports quality of service related to real-time voice services (with guaranteed bandwidth, guaranteed delay, guaranteed packet loss, etc.) as well as security.

A major feature of NGN will be generalised mobility, which will allow consistent provisioning of services to a user, i.e., the user will be regarded as a unique entity when utilising different access technologies, regardless of their types. This feature is also called nomadism.

2.4 A closer Look at the NGN

2.4.1 NGN network elements

NGN makes use of standards based network elements. The following network elements have been introduced by NGN:

- Gateway (GW):
  The GW is responsible for the media stream conversion, i.e. the conversion of TDM based voice signals into IP packets and the signalling protocol conversion. There are three main GW types:
  - Residential Gateway (RG)
    A RG is located at the subscriber's home and provides packetisation of traditional telephone services. It supports a small number of legacy line side interfaces such as POTS or ISDN-BA. Existing telephones can be reused with a RG. Upon receiving the appropriate instructions from the call server (CS), the RG sets up the media path for the voice traffic and performs additional functions such as echo cancellation, compression and tone generation. The RG is connected to the broadband IP access line. A RG is sometimes referred to as integrated access device (IAD).
  - Media Gateway (MG)
    An MG is located at the IP network edge towards the subscriber and provides the

same functions as a RG. An MG connects several hundred subscribers of the residential area it is placed in. Beside the legacy line side interfaces it also supports ISDN primary rate interfaces for TDM PABXs.
The MG is typically deployed within class 5 switch replacement scenarios. A class 5 NGN offers a future-proof network expansion and reduced maintenance costs.

- Trunking Gateway (TG)
  A TG is located between the PSTN network and the packet network. A TG packetises circuit-switched trunks in the PSTN and vice versa. It terminates both the PSTN trunks and packet streams in the packet network, and manages the media connections for the termination of the PSTN trunks as commanded by the call server. It offers no POTS or ISDN line side interfaces, but n x E1 or STM-1 interfaces. By installing TGs as part of new networks, service providers can interconnect existing PSTN TDM networks with the NGN packet networks.

- Call Server (CS) (or softswitch or media gateway controller (MGC))

  The CS provides the control and resource allocation plus overall management of a NGN call. A CS processes the messages received from the GWs using a standardised media gateway control protocol. It also communicates with other CSs to set-up end-to-end calls.

  A CS typically controls several GWs. As this control function is centralised for most services, the CS resides on a powerful server platform.

  A GW normally connects to one primary CS. If GWs are controlled by different CSs, control information is exchanged between the CSs by means of the BICC (Bearer Independent Call Control) or SIP-T (SIP for telephones) protocols.

- Signalling Gateway (SG)

  A SG interconnects the NGN with the PSTN signalling network, i. e. it connects a CS to the signalling network SS7, allowing end-to-end signalling for calls between NGN and PSTN. The SG receives ISUP (ISDN User Part) messages and forwards them to the CS by using the SIGTRAN protocols.

- Application Server (AS)

  An AS implements a software application providing information required by a remote or local application. This application is termed the client. A typical example is an email server that passes email data to an email client on request.

  Other applications can be announcement services or other intelligent network (IN) services.

2.4.2 NGN protocols

The NGN architecture defines four types of protocol categories:

- Transport protocols
  The transport protocols define how the payload (e. g. voice) packets are to be carried over the packet network.
  Examples
  - RTP/UDP/IP (voice)
  - T.38/IFP/UDP/UDPTL/IP (fax)
- Signalling protocols
  The signalling protocols determine how the TDM signalling methods are to be carried in packets, and also how CSs communicate signalling with each other.
  Examples
  - Q.931/SIGTRAN (= IUA/SCTP)/IP
  - BICC
  - SIP-T
- Control protocols
  The control protocols specify how the CS communicates with the MGs.
  Examples:
  - H.248 (from ITU-T), also known as MEGACO (from IETF)/UDP or TCP/IP
  - SIP/UDP or TCP/IP
- Service protocols
  The service protocols constitute the communication between a CS and an AS.
  Examples:
  - CAP
  - Parlay
  - SIP

2.4.3 PSTN replacement

While evolving from the PSTN to NGN, continuity of bearer services should be provided. There are two alternatives addressed within ETSI TISPAN:

- PSTN emulation
  Near-perfect PSTN emulation, with a focus on supporting most if not all legacy PSTN services in a way that is transparent to the end user.
- PSTN simulation
  Simulation of the most popular legacy services and support for the most commonly used set of PSTN services, with possibly different behaviour from some of the services.

PSTN emulation or simulation is based on different bearer services in the NGN environment. The principal difference between PSTN services and emulated or simulated services is primarily Grade of Service (GoS) and QoS properties related to the bearer technology. This is because, for example, the end-to-end one-way delay is typically greater for simulated or emulated bearer services versus the native bearer services.

PSTN emulation

PSTN emulation, as one of the service components of NGN, provides PSTN and ISDN basic and supplementary services and provides IN services. It interworks with the existing networks and other components of NGN.

Therefore, PSTN end users may use existing services and existing terminals under the PSTN emulation environment, without the knowledge of the network being replaced by the NGN. The service control logic and service execution environment is primarily located in a CS. The CS is therefore the responsible network entity for the service delivery.

Highlights for the CS-based PSTN emulation are:

- Provides current PSTN and ISDN services with an unchanged presentation for all interfaces.
- Services and features are provided by the call control of the CS.
- The CS uses the H.248/MEGACO protocol for the control of the media or residential gateways.
- Emulation domain (CS) and IMS session control interwork via SIP.
- Possibility to re-use installed access equipments (POTS, ISDN, PRI, CAS etc).
- Retain existing subscriber base through immediate migration.

PSTN simulation

PSTN simulation refers to the provisioning of PSTN- and ISDN-like services to legacy terminals and advanced terminals such as IP phones. However, there is no guarantee that PSTN simulation would provide all features that have been available to the PSTN user. On the other hand, PSTN simulation may provide additional features and capabilities that have not been available to the users of PSTN.

Contrary to the CS based service control concept of PSTN emulation, the PSTN simulation uses the IP multimedia service component. The application server (AS) houses the service control logic and service execution environment behind CS entities.

Highlights for the IMS based PSTN simulation are:

- Voice service is limited to simulation, with SIP based voice application server.
- Most commonly used and popular voice services are simulated, e. g. CLIP, Call Forwarding etc.
- Focus on new multimedia services.
- Advanced multimedia services like push-to-talk, video conferencing, IP Centrex etc.
- Converged services across different access domains, e. g. mobile networks, broadband wireline networks (DSL, FTTH, etc.), WiFi…
- Interworking with the PSTN emulation domain via SIP.

The figure 1 below shows a high-level presentation of how emulation and simulation is performed and the relationship between different networks and NGN. As shown in the figure, there are several ways how user equipment can be connected to an NGN providing either emulation or simulation of PSTN.

2.4.4 What is IMS?

Introduction to IMS

The IP multimedia subsystem (IMS) is an architectural framework for delivering Internet protocol (IP) multimedia services to end users.

IMS is defined with open standards from 3GPP and ETSI and is based on IETF protocols (SIP, RTP, RTSP, COPS, DIAMETER, etc.). IMS is designed for both wireless and wireline networks and is the basis for fixed and mobile convergence (FMC).

IMS supports operation and interworking with a variety of external networks via defined reference points. Specifically it is capable of interworking with the PSTN.

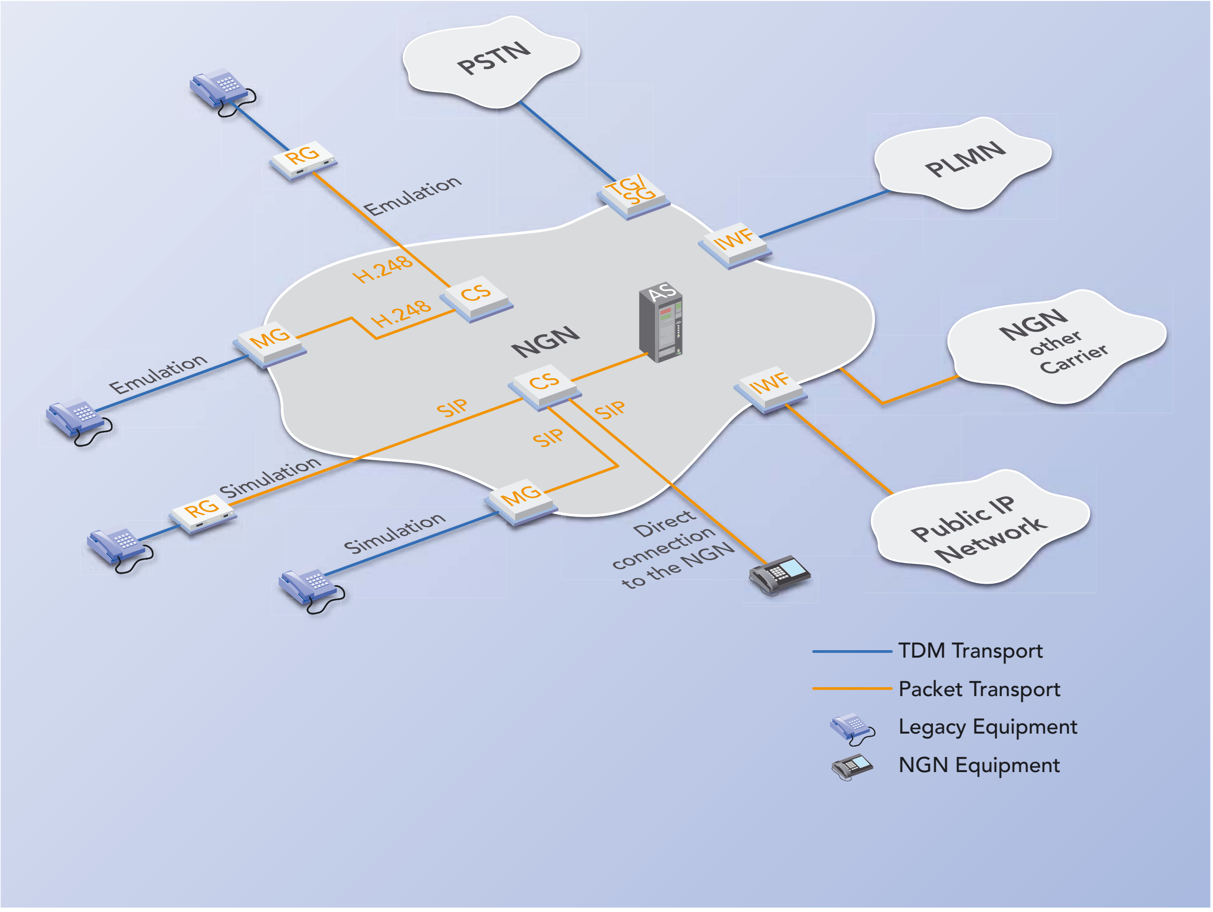

Figure 1: Examples for PSTN emulation and simulation with NGN

IMS features

- IMS delivers IP-based multimedia communications: Person-to-person and person-to-machine.
- IMS fully integrates real-time with non-real-time multimedia communications, e. g. live streaming and chat.
- IMS enables different services and applications to interact, e. g. combined use of presence and instant messaging.
- IMS provides control of the IP connectivity in access networks (QoS, admission control, authentication, etc.).
- IMS enables interworking and interoperability with legacy and other networks.
- IMS is access technology independent for session/call control and applications.

2.4.5 Use of IMS in NGN

In the NGN environment the IMS service component supports the provision of SIP-based multimedia services to NGN terminals. It also supports the provisioning of PSTN simulation services.

In other words, NGN provides the communication network infrastructure while IMS is responsible for the service control.

3 Migration Scenarios

3.1 Evolving from PSTN to NGN

A main target for a network operator is the reduction of parallel communication networks, i. e. TDM for voice and IP for data, to a single network, preferably based on IP transport. This goal can be achieved by extending the IP network reach into the access network area, i. e. by implementing IP based multiservice access nodes (MSAN) close to the subscriber's locations.

The advantages of one common IP network are:

- The cost for the evaluation, commissioning and operation of a single network is lower than for two or more parallel networks.
- Operational costs in IP networks tend to be lower than in PSTN.
- The IP transport network is future-proof.

The IP network is a key component of a NGN. It provides clearly separated transport, control and service layers. The coupling of the different layers is assured through open and standards based interfaces.

An NGN is prepared for future multimedia services based on the IP multimedia subsystem (IMS).

3.2 Migration Steps for Telephony Networks

Having in mind the three main steps as described in the section "Migration principles and procedures":

- optimisation and preparation,
- capacity increase, and
- PSTN replacement,

these main steps are broken down into five more detailed migration steps. The proposed migration procedure primarily targets incumbent local exchange carriers (ILEC) operating fixed networks.

Note that there is no standard procedure fitting every network deployment or service provider demand. There may be many differences in the areas of geographical coverage, network grow rate, aging legacy equipment or demand for new services. The extent and sequence of migration steps can therefore vary and some steps even may be omitted.

Voice migration steps:

Migration	Migration step	Network
Start	Existing networks	PSTN for TDM voice and narrowband Internet access and broadband Internet access via DSLAMs.
Step 1	PSTN consolidation	Introduction of high capacity local and trunk exchanges, replacement of DSLAM and DLC network elements by integrated MSANs.
Step 2	NGN in the core network	Replacement of trunk exchanges (class 4 switches) by class 4 call servers and trunking gateways (TG).
Step 3	NGN in the residential area	Deployment of residential gateways to connect to legacy telephones, and CPE replacement with soft clients or IP phones.
Step 4	NGN in the access network	Replacement of local exchanges (class 5 switches) by class 5 call servers and media gateways. Media gateways can be integrated into MSANs.
Step 5	Shutdown of PSTN	Replace remaining TDM exchanges

3.3 Existing Networks: PSTN for TDM Voice and Internet Access

The public switched telephone network (PSTN) is a hierarchical network built around call-processing and routing units known as exchanges. The PSTN provides carrier-grade quality with well defined QoS criteria and standardised engineering rules.

Local exchanges (LE), also known as class 5 switches, provide switched services to the subscriber device, e. g. a telephone set. A LE commonly handles 10,000s to 100,000s of lines depending on whether it is a rural or urban switch.

Trunk exchanges (TE), also called class 4 switches, provide routing/switching between LEs. Arranged in a hierarchical architecture, there can be two levels of TE: Primary TE and secondary TE. There is typically 1 secondary TE to every 10 LE, and 1 primary TE to every 6 or 7 secondary TE. There is also a small number of TEs used as international gateways, about 1 for every 2 primary TE.

In this network, all voice traffic is transported over TDM. The digital loop carrier nodes (DLC) perform the concentration of the subscriber's voice traffic to higher capacity links.

The voice related signalling is handled by the SS7 signalling network. The signalling transfer point (STP) provides the transfer of signalling messages. A service control point (SCP) implements the intelligent networks (IN) value-added services.

With the adoption of Internet usage, local exchange operators provide connectivity to Internet service providers (ISP) to their subscribers.

In a communication network the Internet access is provided either through narrowband dialup services using the inband bearer channel, or through broadband DSL, using splitters to separate the voice signal from the data signal.

The data gateway between the PSTN and the IP network is a narrowband access server (NAS). The DSL line is connected to a DSLAM and is terminated in a broadband remote access server (BRAS).

In general telecom service providers offer their subscribers DSL for broadband data access. Due to the limited reach of DSL, the digital subscriber line access multiplexers (DSLAM), terminating the digital signal at the network side, must be brought closer to the subscriber.

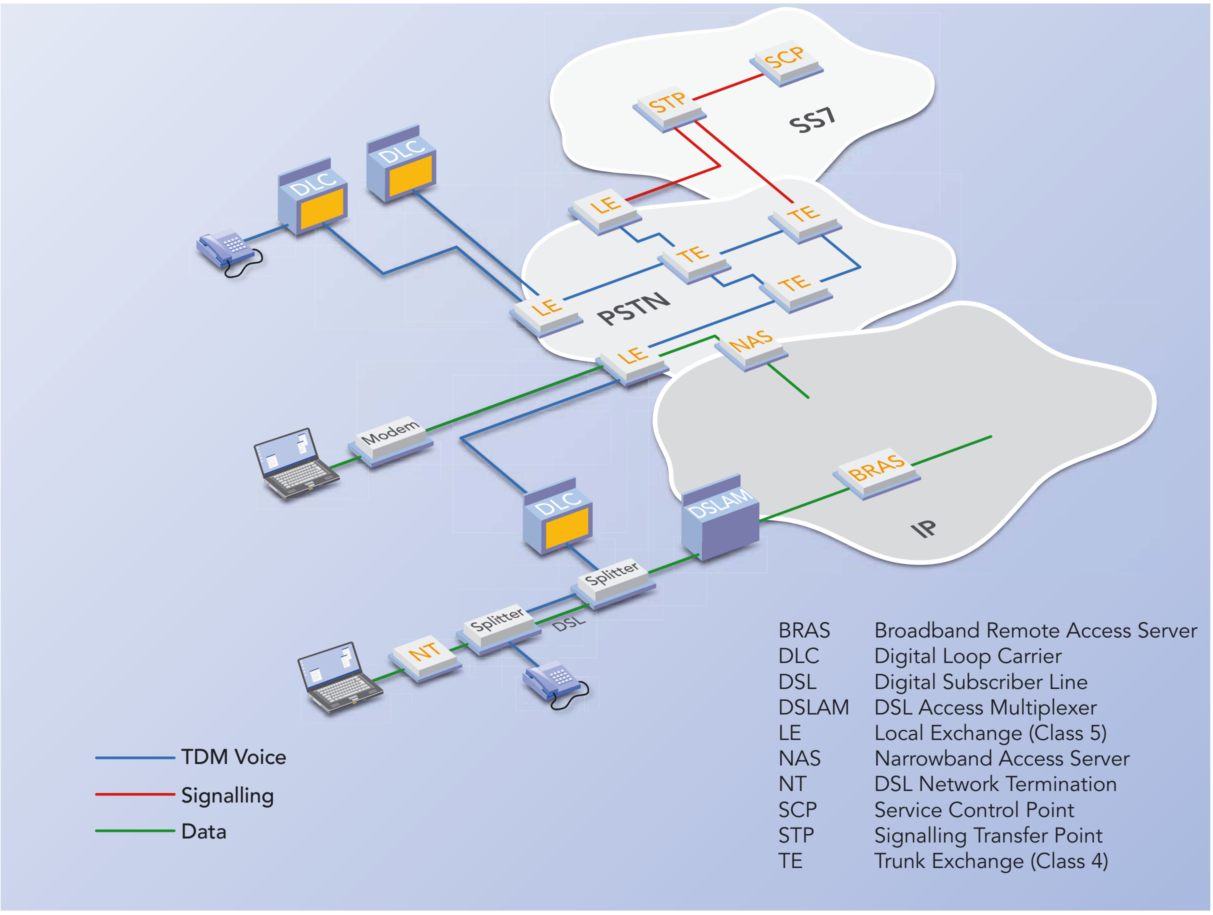

Figure 2: PSTN and IP network architecture for voice and data applications

3.4 PSTN Consolidation: High Capacity Exchanges

With the availability of higher capacity local exchanges and trunk exchanges, the PSTN infrastructure is consolidated and optimised by replacing several lower capacity exchanges by higher capacity exchanges.

The reduction of LEs implies the widespread deployment of DLCs to keep the wire length in the local loop under control. The network infrastructure optimisation reduces the service providers' operational cost.

Separate DLCs and DSLAMs can be replaced by MSANs incorporating the functions of both network elements. Data bandwidths offered to the subscribers can be increased by bringing the MSAN closer to the subscriber's location.

Note that in this scenario voice is still a pure TDM application.

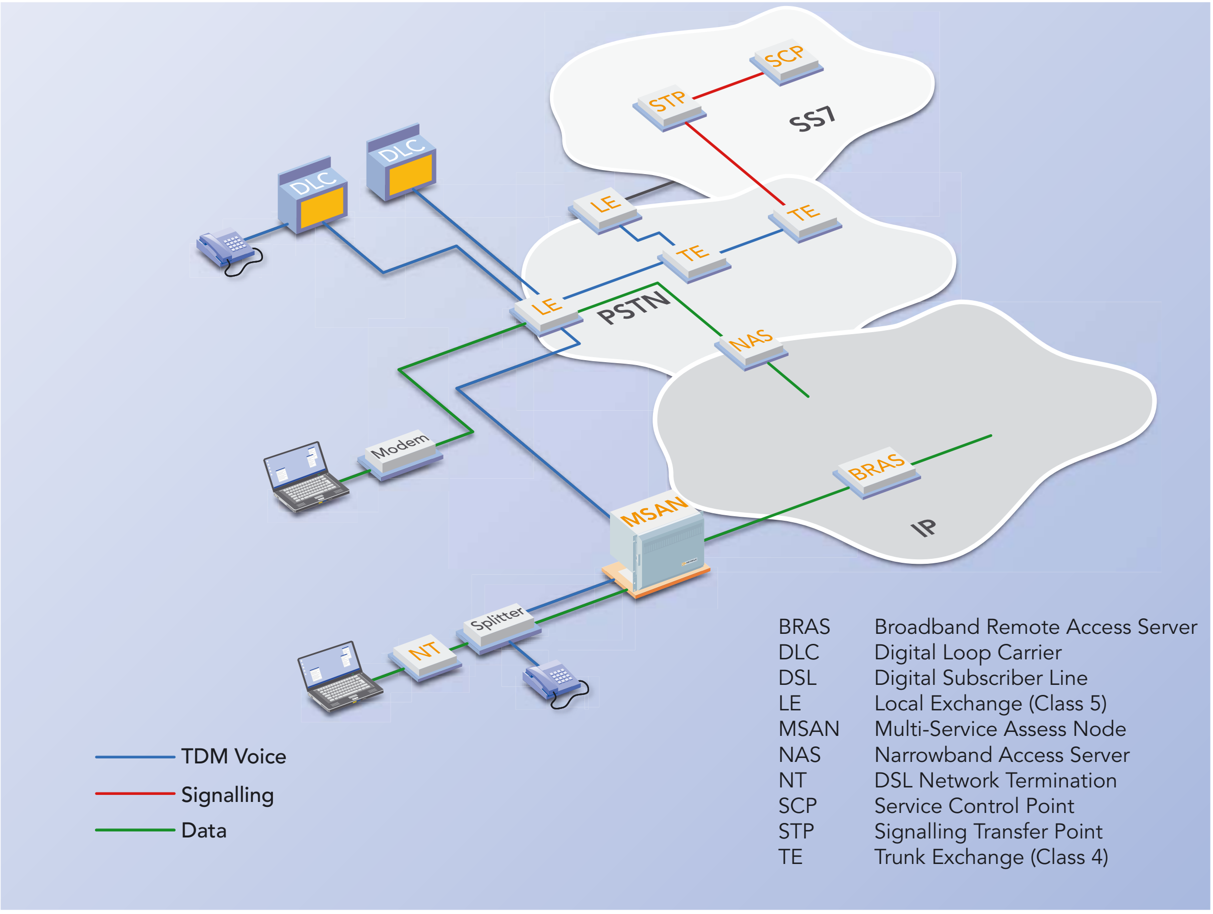

Figure 3: PSTN and IP network architecture for consolidated voice and data applications

When deploying an MSAN, a network operator has to keep in mind that he will migrate this MSAN to NGN one day. A product offering the reuse of the POTS, ISDN and DSL line cards will be preferred.

In an MSAN prepared for NGN, the TDM voice part can be upgraded to VoIP by simply replacing or even upgrading the protocol conversion unit (e. g. V5.2) to an media gateway. All other units can be reused. The TDM uplink unit will not be used anymore.

An MSAN upgrade is a fast and cost effective way of the migration to NGN in the access area.

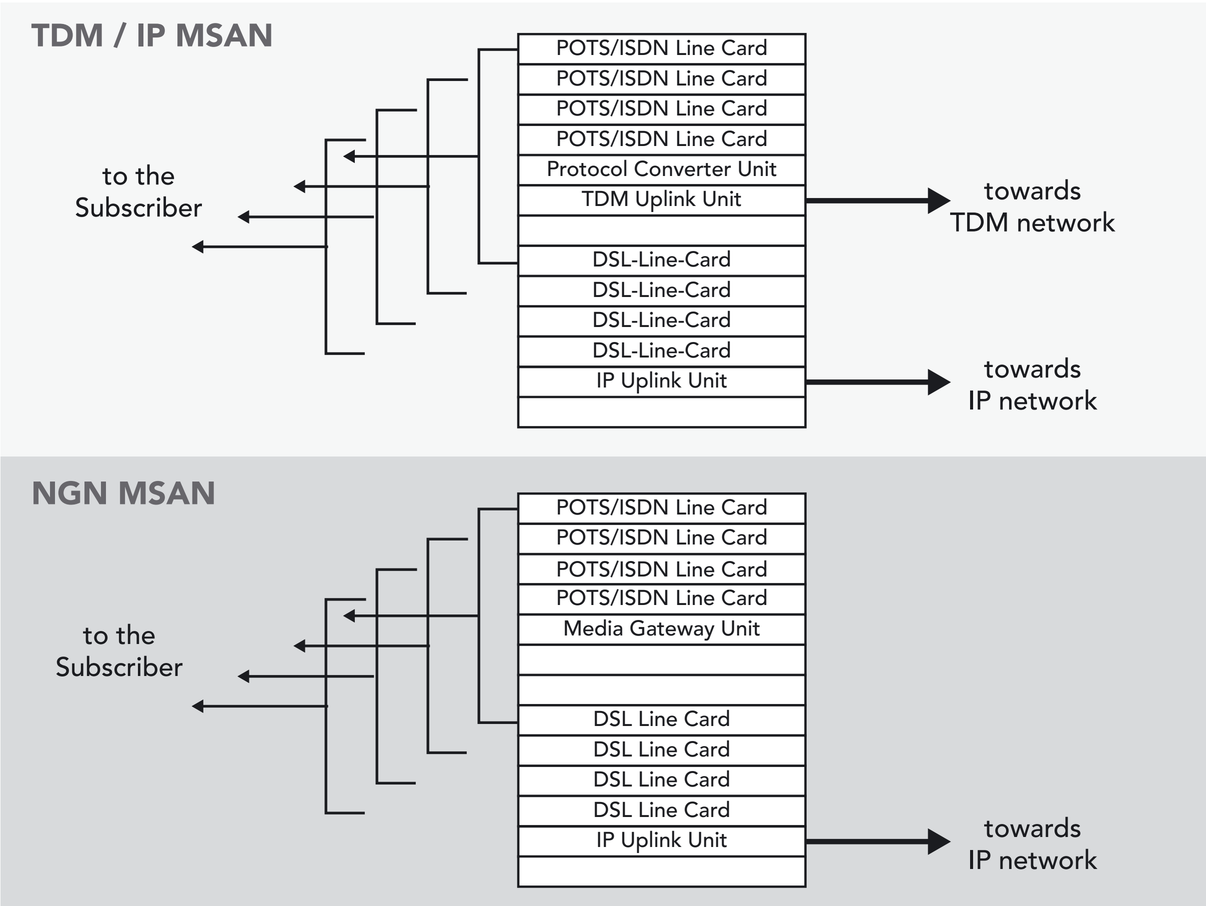

Figure 4: MSAN migration with reuse of the POTS/ISDN line cards, the DSL line cards and the IP uplink unit

3.5 NGN in the Core Network: Class 4 Switch replacement

Replacing the trunk exchanges (TE, class 4 switch) at the core of the PSTN by a class 4 call server and trunking gateways (TG) is the first step towards a true NGN. TGs are mainly used to offload long distance calls from the PSTN. The PSTN edge, i. e. the local exchanges, is not touched.

A Trunking Gateway

- emulates a TE and the value added services.
- replaces a TE or increases the traffic handling capacity of a TE.
- interfaces to the LE and TE on the PSTN side.
- communicates with a class 4 call server through H.248/MEGACO protocols.

Implementing the PSTN emulation service will keep the subscribers unaware of the migration to NGN.

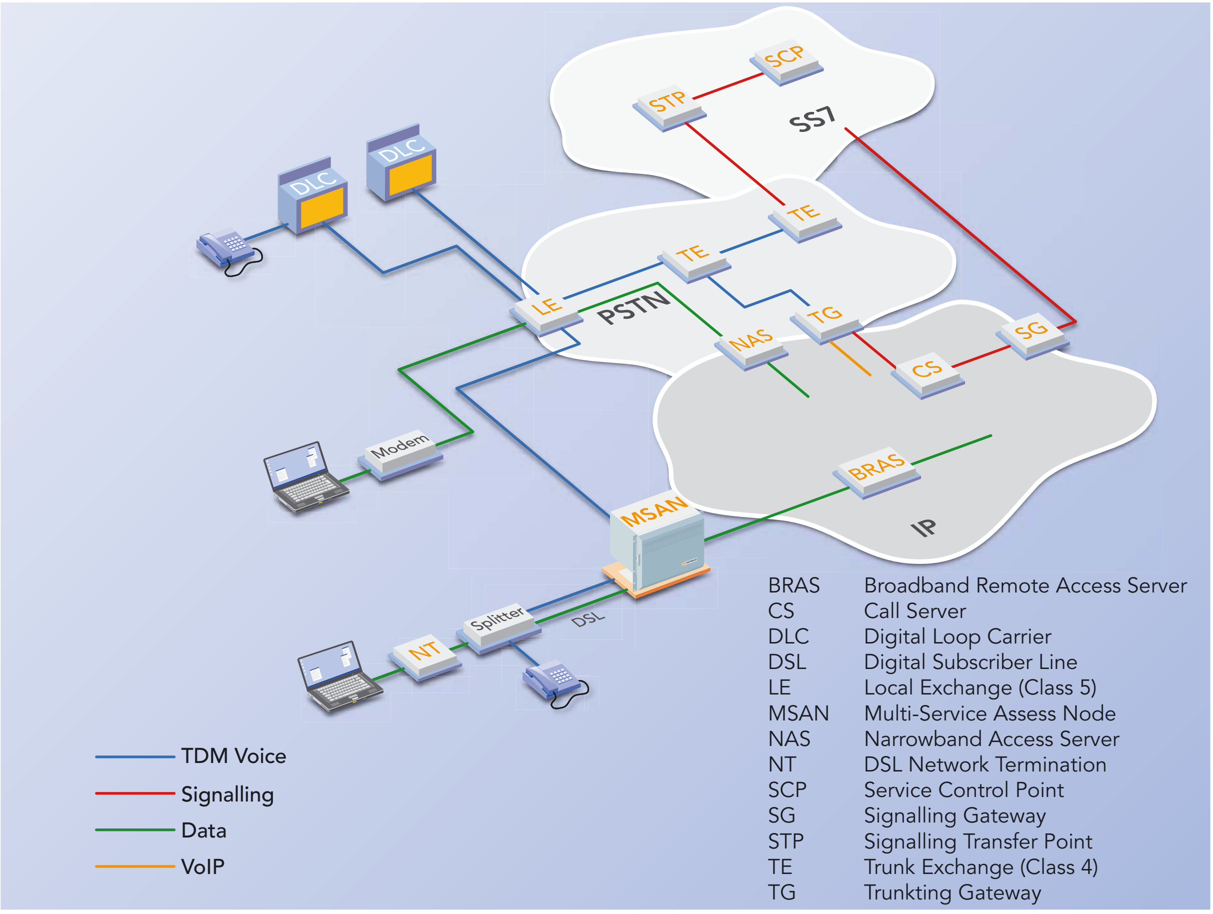

Figure 5: PSTN and IP network architecture using trunking gateways

3.6 NGN in the Residential Area: CPE replacement

The introduction of broadband access networks enables the deployment of voice and multimedia IP clients at the subscriber's location. A subscriber using an IP phone or a legacy phone connected to a residential gateway (RG) disposes of a pure IP access towards the telecommunication network.

The subscriber can be connected to a DSLAM, handling pure IP traffic, or he can be connected to a MSAN offering new multimedia services. These new services will allow service providers to differentiate and compete with alternative carriers.

It is assumed that voice service is implemented as PSTN simulation, using SIP instead of H.248/ MEGACO as the control protocol. Using SIP also enables the seamless evolution towards the IMS.

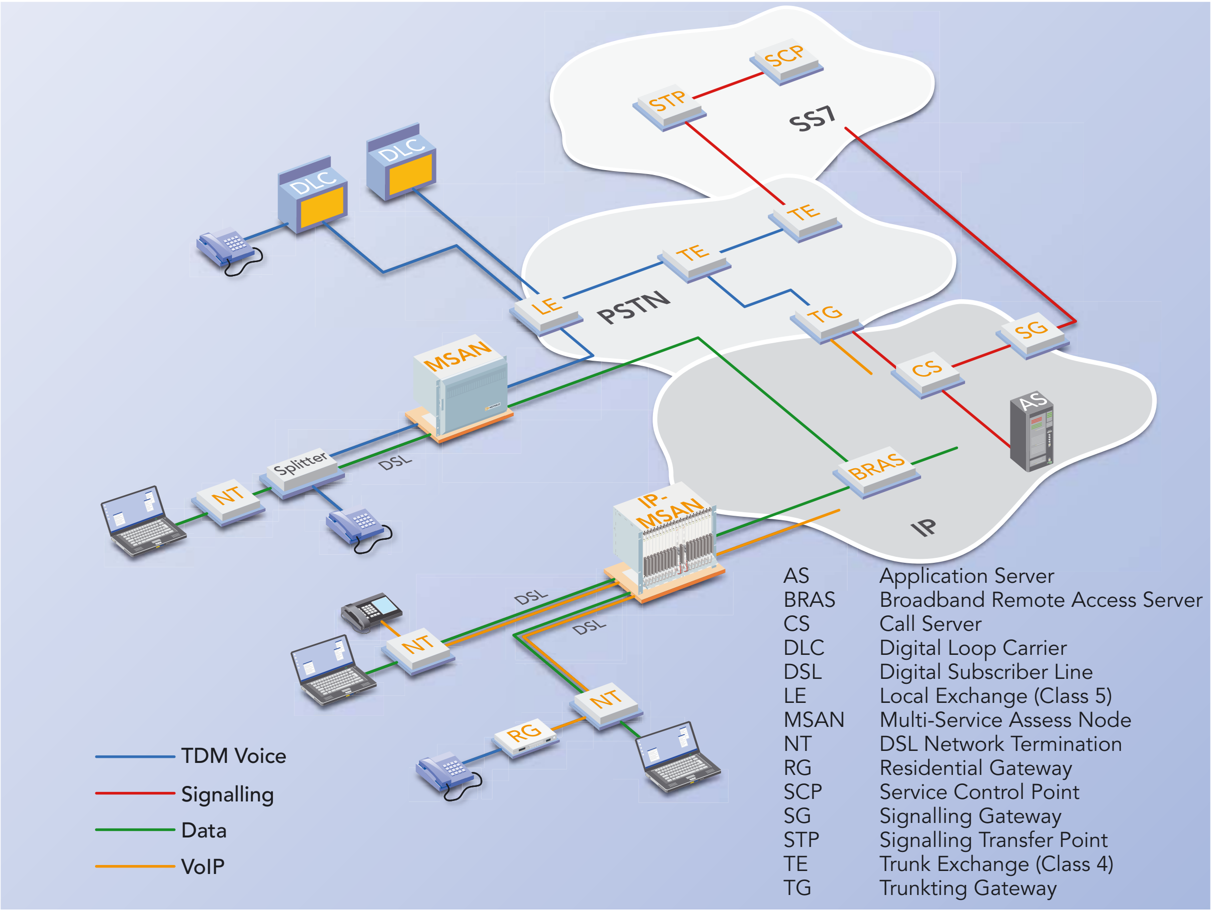

Figure 6: PSTN and IP network architecture using residential gateways and IP phones

3.7 NGN in the Access Network: Class 5 Switch replacement

Class 5 call servers and media gateways (MG) replace local exchanges (LE, class 5 switch) at the edge of the PSTN. This extends the NGN closer to the subscriber's location. MGs are mainly used to increase the network capacity or to replace obsolete LEs.

A Media Gateway

- emulates a LE, its supplementary services and optionally the value added services.
- replaces a LE or increases the traffic handling capacity of the access network.
- interfaces to the subscribers on the PSTN side.
- communicates with a class 5 call server through H.248/MEGACO protocols.

An MG can be deployed as a separate network element or be integrated into an MSAN, offering also DSLAM and multimedia functionality. Implementing the PSTN emulation service will still keep the subscribers connected to an MG unaware of the migration to NGN, as legacy interfaces (POTS, ISDN) are provided over the MSAN.

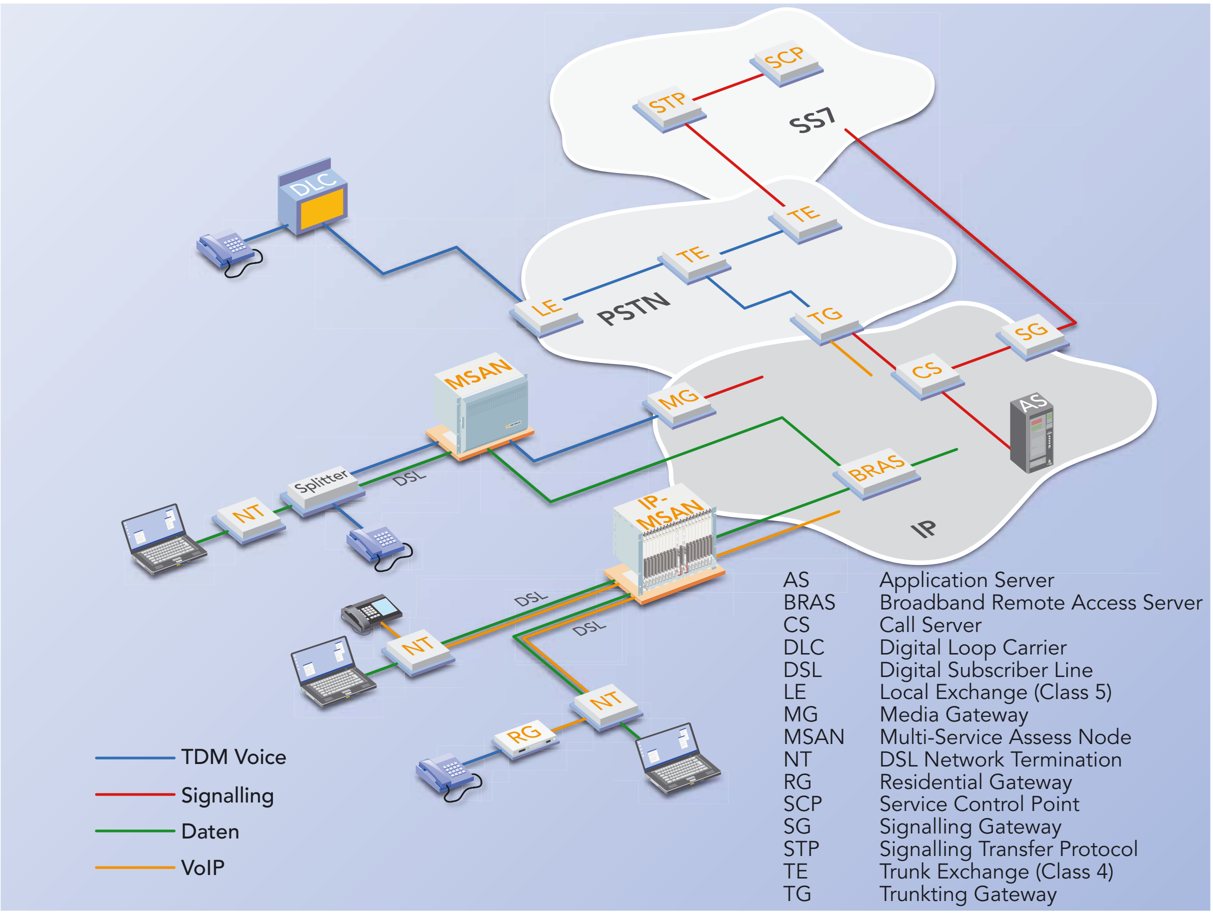

Figure 7: PSTN and IP network architecture with Media Gateways

3.8 Shutdown of PSTN

As a final migration step towards the all-embracing NGN, the remaining legacy PSTN equipment is replaced by NGN network elements.

The target for this last step is the replacement of obsolete equipment, i.e. equipment reaching end-of-life, and cost reduction by having one single network to maintain and operate.

Note that also in this scenario subscribers still can use their legacy equipment which is connected to a residential gateway or to a media gateway.

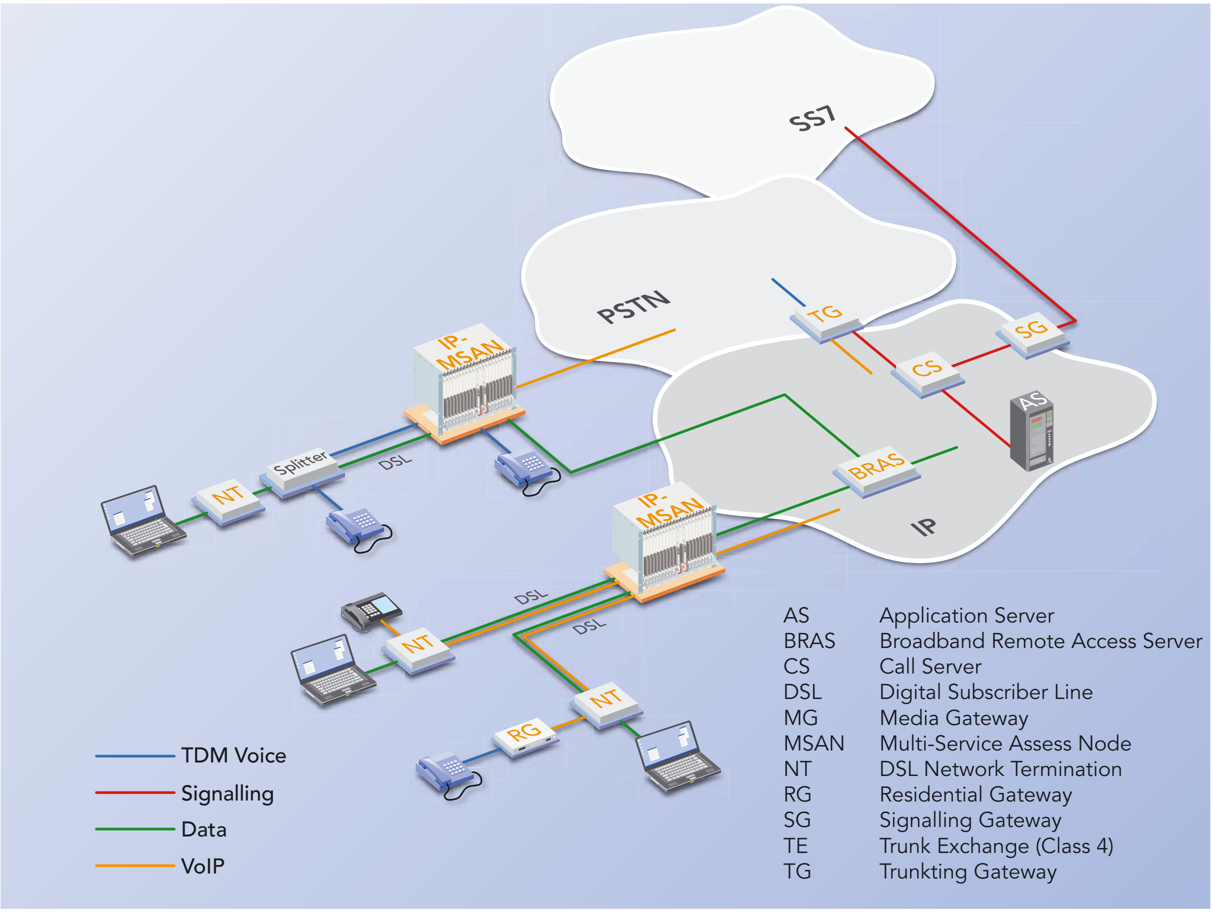

Figure 8: IP network architecture using IP phones, residential gateways and MSAN integrated media gateways

4 How KEYMILE supports Voice Migration

KEYMILE is a leading manufacturer and supplier of next generation access systems with a long history in providing voice applications supporting POTS, ISDN and V.5x services. Its product spectrum covers IP based DSLAMs and TDM or IP based MSANs.

KEYMILE products are used as access platforms and allow seamless conversion from TDM towards packet-oriented services. They combine carrier-grade broadband access, telephony and data interfaces in a single compact next generation access platform with effective IP-DSLAM functionality. While supporting legacy telephony with POTS, ISDN, V5.x and broadband services, the MSANs can smoothly be upgraded to the next generation network based on H.248/MEGACO or SIP protocols. Concurrently they provide high quality Triple Play and broadband business services.

KEYMILE serves customers in over 100 countries with subsidiaries worldwide and a global network of partners.

5 Summary

5.1 Step by Step Migration

The PSTN will not be replaced by the voice NGN from one day to another. There will be a stepwise replacement of obsolete PSTN network elements or a capacity expansion, introducing NGN network elements, as

- trunking gateways in the network core,
- residential gateways at the subscribers home,
- media gateways at the network edge, together with
- call servers controlling the deployed gateways.

Voice will be converted from TDM to IP first in the core network. The TDM to IP conversion will gradually continue at the subscribers location and finally expand to the edge of the network.

The migration can be driven by cost, e.g. OPEX reduction, driven by increased capacity demand or by the service providers aiming to offer new (multimedia) services to their customers. In any case the specific factors influencing the migration must be considered carefully by the service provider.

NGN in the access network is tightly coupled with the deployment of broadband data access. Digital loop carriers (DLC) and digital subscriber line access multiplexers (DSLAM) will converge to multi-service access nodes (MSAN) offering the functionality of a DLC, DSLAM and media gateway in one network element. MSANs will be placed close to the subscribers homes to be able to deliver the high data rates required for multimedia services.

5.2 NGN is not only Voice

NGN not only delivers voice service but enables the deployment of multimedia services. Multimedia services using the IMS are based on a NGN infrastructure. NGN is therefore a precondition to IMS.

The focus of this white paper lies on the essential elements of a migration from TDM voice to VoIP, mainly in the wireline access area. There are many other factors in the access or core network influencing the migration strategy. Among them are:

- Revenue from new services
  - Value-added NGN voice
  - Multimedia services
- Segmentation and services
  - Enterprise voice, VoIP and hosted Centrex services
  - Enterprise IT outsourcing
- Detailed CAPEX evaluation
  - Cost of NGN core network elements
  - Cost of converged multi-service access network elements (MSAN)
  - Cost of bringing fibre closer to the subscribers
- Detailed OPEX model
  - Staff training for new network elements
  - Increased efficiency through convergence

6 Abbreviations

Abbreviation	Meaning
3G	Third Generation mobile network or service
3GPP	Third Generation Partnership Project
AS	Application Server
BB	Broadband
BICC	Bearer Independent Call Control
BRAS	Broadband Remote Access Server
CAMEL	Customized Application for Mobile network Enhanced Logic
CAP	CAMEL Application Part
CAPEX	Capital Expenditure
CAS	Channel Associated Signalling
CLIP	Calling Line Identification Presentation
COPS	Common Open Policy Services
CPE	Customer Premises Equipment
CS	Circuit Switched
CS	Call Server
DIAMETER	DIAMETER is a protocol intended for access, authorization and accounting support
DLC	Digital Loop Carrier
DSL	Digital Subscriber Line, e. g. ADSL or VDSL
DSLAM	Digital Subscriber Line Access Multiplexer
E1	First level of PDH, 2048 kbit/s
ETSI	European Telecommunications Standards Institute
FMC	Fixed and Mobile Convergence
FTTx	Fibre-to-the-x, x = Home or Curb or …
GoS	Grade of Service
GPRS	General Packet Radio Service
GSM	Global System for Mobile communication
H.248	Media gateway control protocol, identical to MEGACO
IAD	Integrated Access Device
IETF	Internet Engineering Task Force
IFP	Internet Facsimile Protocol (T.38)
ILEC	Incumbent Local Exchange Carrier
IMS	IP Multimedia Subsystem
IN	Intelligent Network
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ISDN-BA	ISDN Basic Access
ISDN-PRA	ISDN Primary Rate Access
ISP	Internet Service Provider

Abbreviation	Meaning
ISUP	ISDN User Part
IT	Information Technology
ITU-T	International Telecommunications Union – Telecommunication standardization sector
IUA	ISDN Q.921-User Adaptation layer
IWF	Interworking Function
LE	Local Exchange (class 5 switch)
MEGACO	Media Gateway Control protocol
MG	Media Gateway
MGC	Media Gateway Controller
MPLS	Multi Protocol Label Switching
MSAN	Multi Service Access Node
NAS	Narrowband Access Server
NGN	Next Generation Network
OPEX	Operational Expenditure
PABX	Private Automatic Branch Exchange
PC	Personal Computer
PDH	Plesiochronous Digital Hierarchy
PLMN	Public Land Mobile Network
POTS	Plain Old Telephone Service
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
Q.931	ISDN connection control protocol
QoS	Quality of Service
RG	Residential Gateway
RTP	Real-time Transport Protocol
RTSP	Real-Time Streaming Protocol
SCP	Service Control Point
SCTP	Stream Control Transmission Protocol
SDH	Synchronous Digital Hierarchy
SG	Signalling Gateway
SIGTRAN	Signalling Transport (= IUA / SCTP)
SIP	Session Initiation Protocol
SIP-T	SIP for Telephones
SLA	Service Level Agreement
SS7	Signalling System #7
STM-1	First level of SDH, 155'520 kbit/s
STP	Signalling Transfer Point
TDM	Time Division Multiplexing
TE	Trunk Exchange (class 4 switch)
TG	Trunking Gateway
TISPAN	Telecoms & Internet converged Services & Protocols for Advanced Networks

Abbreviation	Meaning
TV	Television
UDP	User Datagram Protocol
UDPTL	UDP Transport Layer
V5.x	V5.1 or V5.2
VoBB	Voice over Broadband
VoIP	Voice over IP

Publisher

KEYMILE GmbH
Wohlenbergstraße 3
30179 Hanover, Germany

Phone +49 511 6747-0
Fax +49 511 6747-450
Internet www.keymile.com
Mail info@keymile.com

