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Telecom Regulatory Authority of India

Voice over LTE

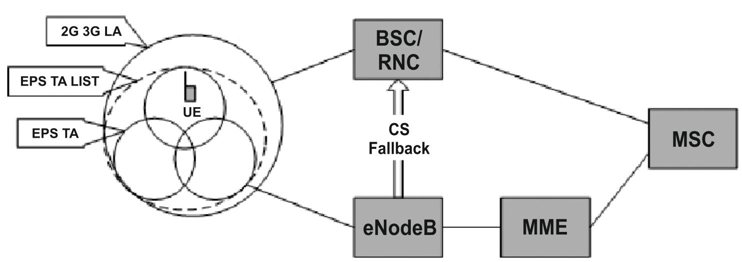
LTE technology is becoming popular as the next generation technology supporting high data rates. It is the next step in the evolution of cellular communication data networks. The LTE radio access has been designed for IP-based services. This means that LTE has no support for Circuit Switched (CS) channels optimized for voice calls. It is a packet-access only which is not compatible to the circuit-switched networks. This is different from GSM, WCDMA and CDMA, which support both circuit- and packet-switched services.

As voice call application has been the primary application since the inception of cellular networks, how much ever the data services become popular, users still expect voice as a basic service provided by the operator. As a result the support for Circuit Switched (CS) raises the question of how to provide the voice call over LTE and how to ensure the voice call continuity when the user moves from LTE to 2/3G networks. Over the last few years, there have been multiple views, technologies, and evolution paths for LTE voice services. Today the following solutions have emerged as possible commercial choices:

- Circuit Switched Fallback (CSFB)
- Voice over IMS, Single Radio Voice Call Continuity (SRVCC)
- Multimedia Telephony (MMTel) over LTE
- Over-the-top (OTT) like Skype
- Voice over LTE via Generic Access (VoLGA)

Circuit Switched Fallback (CSFB)

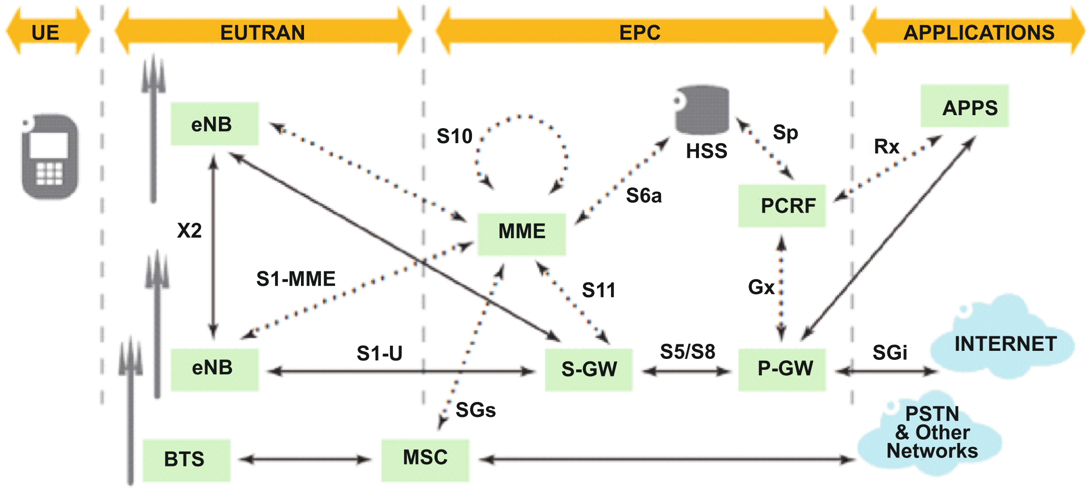
CSFB is the mechanism to move a subscriber from LTE to a legacy technology to obtain circuit switched voice service. This function is only available if LTE coverage is overlapped by either GERAN (GSM EDGE Radio Access Network) coverage or UTRAN (UMTS Terrestrial Radio Access Network) coverage as shown in figure 1



LA : Location Area
EPS : Evolved Packet System
TA : Tracking Area
RNC : Radio Network Controller
MME : Mobile Management Entity

Figure 1 Overlapping 2G/3G LA and EPS TA for CS fallback

Evolved Packet System (EPS) is the architecture of the LTE cellular standards .EPS encompasses the Evolved UMTS Terrestrial Radio Access Network (E-UTRAN) and Evolved Packet Core (EPC) that promises to deliver higher throughput, lower latency, and larger bandwidth over a simple, flat IP architecture. EPS is responsible for IP layer interception of Content of Communication (CC) data. In addition to CC data, the Lawful Interception (LI) solution for EPS offers generation of Intercept Related Information (IRI) records from respective control plane (signalling) messages as well. EPS architecture for CS fallback is shown in figure 2.



PCRF : Policy and Charging Rules Function
S-GW : Serving Gateway
P-GW : Packet Data Network Gateway
HSS : Home Subscriber Server

Figure 2 EPS architecture for CS fallback

CS fallback is realized by using the SGs interface between the Mobile Switching Center (MSC) Server and the Mobile Management Entity (MME) which is the key control node for the LTE network and efficiently performs a combined registration when the user terminal attaches to the LTE network. SGs interface is used for the mobility management and paging procedures between EPS and CS domain, and is based on the Gs interface procedures. The Gs interface is the reference point that exchanges signaling with MSC, which connects to Serving GPRS Support Node (SGSN). The serving GPRS support node (SGSN) is responsible for routing the packet switched data to and from the mobile stations (MS) within its area of responsibility.

The user equipment (UE) sends an Extended Service Request (CS fallback Indicator) to MME. Extended service Request message is encapsulated in radio resource control (RRC) and S1 Application Protocol (S1 AP) messages. RRC is the layer 3 control plane protocol over the LTE air interface, terminating in the UE and in the base station (termed eNode B in LTE). S1 AP is the signaling service between E-UTRAN and the Evolved Packet Core (EPC) that fulfills the S1 Interface functions such as Initial context transfer function, Mobility functions for UE, Paging, Reset functionality, NAS signaling transport function, Error reporting, UE context release function and Status transfer.

CS fallback Indicator indicates MME to perform CS fallback. The MME sends an S1 AP Request message to eNB that includes a CS fallback indicator. This message indicates eNode B that the UE should be moved to UTRAN/GERAN, the eNode B triggers PS handover to a GERAN/UTRAN neighbour cell by sending a handover Required message to the MME. After the successful handover from PS to CS domain, a normal CS call establishment procedure follows.

If the user has an active LTE data session when a voice call is initiated, the data session may be handed over to the GSM/UMTS network or dropped, depending on the characteristics of the network. If the fallback network is UMTS, the E-UTRAN will perform a packet-switched (PS) handover (PSHO), enabling the data session to continue during the duration of the voice call. However, if the fallback network is GSM, a PSHO may only occur if the GSM network as well as the user device supports Dual Transfer Mode (DTM), which enables voice and data calls to be handled simultaneously. Otherwise, the data session is suspended for the duration of the voice call.

CS fallback extends the life of the GSM/ UMTS network by enabling it to provide voice services for the LTE network. GSM/ UMTS components such as MSCs, CS service platforms, operations support systems (OSSs), and prepaid /post-paid billing systems are all reused, ensuring a fast and quality rollout of voice services for LTE. No new network elements need to be added, and required upgrades to existing network nodes are relatively minor compared to the other options. CS fallback changes to the MSC are not complicated because the SG s interface was purposely based on the Gs interface that is currently used between the MSC and Serving GPRS Support Node (SGSN). Handsets reuse the GSM/ UMTS client with only a few added enhancements for CS fallback.

Conversely, CS fallback is quite signaling-intensive and fallback may take a while to complete, with estimates placing it at about 500 ms. In addition, this delay may be increased if the mobile device must conduct measurements to find a suitable GSM/UMTS cell to use and must then perform a location update before being able to originate or answer a call. This call set up delay may be enough to be noticed by some LTE subscribers.

Single Radio Voice Call Continuity (SRVCC)

SRVCC is a 3GPP standardized feature which addresses the issue of handover between IMS-based PS domain and a CS domain. 3GPP has standardized SRVCC in Release 8 (2008). SRVCC procedure takes place when a single radio User Equipment (UE) accessing IMS-anchored voice call services switches from the LTE network to the Circuit Switched (CS) domain while it is able to transmit or receive on only one radio access network at a given time. This eliminates the need for a UE to have simultaneous multiple Radio Access Technology (RAT) capability. RAT indicates the type of radio technology to access the CN (Core Network) for example UTRA (UMTS Terrestrial Radio Access), cdma 2000 and GERAN. Figure 3 shows the concept of SRVCC and its reference architecture. When a UE detects movement from UMTS to LTE radio access, it informs, the LTE network via the scheduled measurement gaps (Measurement gaps are periods when no uplink or downlink transmissions are scheduled so that the UE may perform the measurements). The Mobility Management Entity (MME) in conjunction with an IMS network performs handover preparation through signaling. A new call is established between Mobile Switching Centre (MSC) and the IMS network before the old call leg is released. After call transfer to the circuit switched domain is successfully completed, the MSC informs the MME of the successful completion. In turn, the LTE network informs the UE to tune to the UMTS radio access network. The call proceeds in UMTS without discernable interruption.

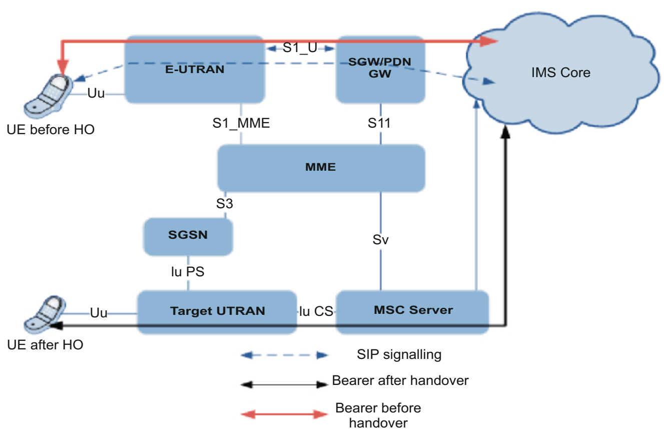


Figure 3 SRVCC reference architecture

SRVCC requires additional enhancements to be made to supporting nodes in UMTS and IMS networks in order to support smooth handover. In UMTS, the MSC Server is deployed alongside the MME in LTE via a new interface known as the Sv interface. This interface enables communication between the MME and MSC. The MSC is responsible for the control of mobile-originated and mobile terminated CS calls. It receives the user-network signaling and translates it into the relevant network to network signaling, e.g. from SIP Signaling in IMS to SS7 signaling in CS domain. In IMS, the Application Server (AS) is enhanced to support SRVCC by enabling centralized IMS services and executing session transfers between the user and the IMS network.

SRVCC enables seamless handover between E-UTRAN and GERAN/UTRAN domains, thus improving the user experience and QoS for Voice over LTE. The user experience of IMS telephony is at same level as with 2G/3G telephony, since involuntary call disconnection is eliminated. SRVCC does not mandate any solution (e.g. MMTel) for voice over LTE. However, it complements any IMS based solution for Voice over LTE with PS to CS handover possibility.

Takeover TV: Facilitating the social negotiation of television content in public spaces

Takeover TV is a platform that allows public displays to become aware of the content preferences of nearby users, allows these users to visualize the collective preferences of those present, and to start and participate in votes that determine what is shown on the display.

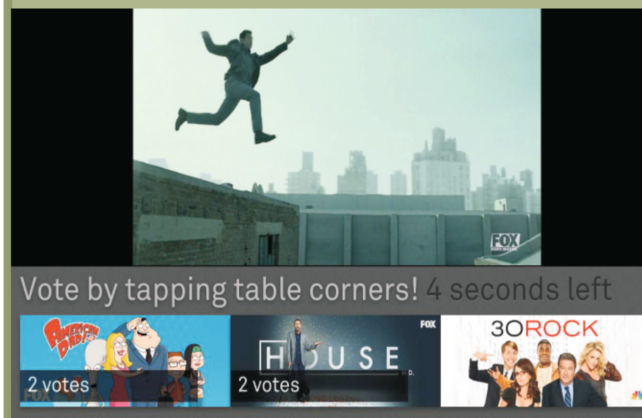


Figure Takeover TV large display interface while a vote is active

When a user arrives at a location, their likes and dislikes automatically influence what can be shown on local displays. Those that want more control can start a vote to choose a new show using their phone. The system supports a continuum of lean-forward and lean-backward interaction strategies, allowing public displays to scale from passive broadcasting (current model) all the way up to full social negotiation where all users vote and interact via mobile phones and augmented tables to decide what should be shown and when.

Source: IEEE

Multimedia Telephony (MMTel)

MMTel was first introduced as a concept in 3GPP release 7 (R7), which was completed in December 2007 and has been further enhanced in subsequent 3GPP releases. Focusing on fixed access, European Telecommunications Standards Institute (ETSI) and its working group, Telecoms & Internet converged Services & Protocols for Advanced Networks (TISPAN), initiated work to standardize multimedia telephony in 2004. TISPAN endorsed IP Multimedia Subsystem (IMS) as the core network, referring to 3GPP specifications for session control and basic communication. MMTel is a service set that uses IMS architecture. IMS is standardized architecture for controlling and delivering multimedia services that employ IP for transport and Session Initiation Protocol (SIP) for service signaling. MMTel standard defines both NNI and UNI, making it acceptable for the mass market.

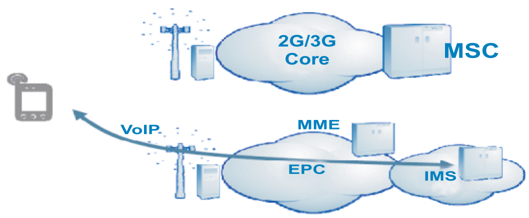


Figure 4 Architecture of MMTel

Based on the IP Multimedia Subsystem (IMS), MMTel offers converged fixed and mobile, real-time multimedia services that allow users to communicate using voice, video and chat. Users can also share image files and video clips easily. MMTel combines quality, interoperability, efficiency, regulatory and supplementary services with rich media and dynamics of internet community-based communication. The intent of MMTel is to eventually phase out fixed and mobile circuit-switched technologies, replacing them with an all-IP solution.

Standardisation of interworking between the MMTel voice component and PSTN/ISDN/PLMN Circuit switched voice was initiated in 2005. At Present the interworking specification from 3GPP and TISPAN provide a stable framework.

3GPP also has standardised interworking between MMTel voice/video and circuit-switched video telephony as used in other 3GPP networks. Figure 5 shows the standardised interworking comprising the following scenarios:

- MMTel speech (duplex)-PLMN/PSTN speech
- MMTel speech +video (duplex)-PLMN video telephony
- MMTel speech +media sharing-PLMN CSI (CAMEL Subscription Information) based media sharing

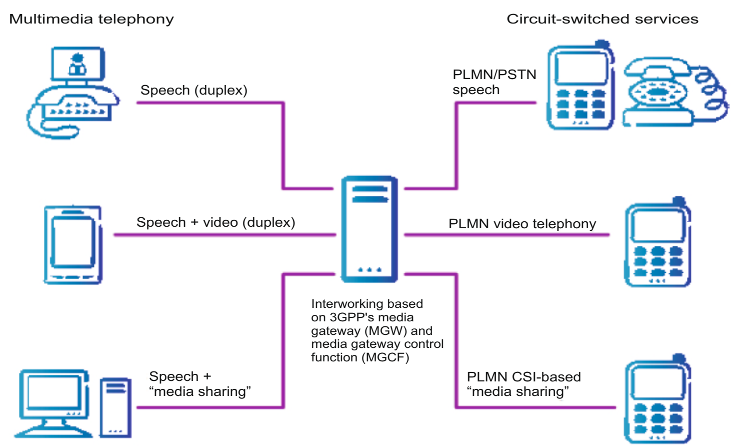


Figure 5 Interworking scenarios between MMTel and standard CS services

Because MMTel is based on IMS, operators can perform network consolidation, which lowers capital and operational expenditure (capex and opex), and MMTel gives operators a host of new multimedia services in other words, opportunities to tap into new revenue streams. With features like network-centric presence-enabled address book, chat, image sharing, video sharing, and file sharing, IMS and MMTel can compete with rapidly expanding internet communities and services, such as MSN and Skype. It also allows operators to remain service-aware, which includes taking part in service delivery and charging. The standardized NNI enables operators to interconnect with one another, creating the potential for truly global, mass-market acceptance and profitability.

MMTel aims to maintain service quality by standardizing the minimum performance for voice and supporting the 3GPP-standardized voice and video codecs. The MMTel services are wide-ranging and include supplementary services, regulatory services, and services that did not exist previously within PSTN/ISDN.

Over-the-top (OTT)

Over-the-top (OTT) services, using applications like Skype and Google Talk are one of the solutions to provide voice call over LTE. LTE boasts features like broad bandwidth, low latency, being always-online, and All-IP, creating natural convenience for the development of OTT and making OTT voice calls almost barrier-free. However, we should also note that now and in the foreseeable future, the voice call service is, and will still be, the main revenue source for the mobile operators. So handing the LTE voice service over completely to the OTT actors is thus something which is expected to not receive too much support in the telecom industry.

Compared to OTT, telecom operators have their own unique advantages in providing LTE voice calls, such as user ID resource, standard-based interconnection, QoS assurance, the ability of handover to CS, and tariff package bundled with data service. In the future, the percentage of OTT calls may increase drastically, especially for the long-distance call. However, the call service provided by telecom operators will still be the mainstream for a long time.

Voice over LTE via Generic Access (VoLGA)

VoLGA is based on the existing 3GPP Generic Access Network (GAN) standard. The purpose of GAN is to extend mobile services over a generic IP access network. VoLGA involves connection of the already existing Mobile Switching Centre (MSC) to the LTE network via a gateway (VANC-VoLGA Access Network Controller).

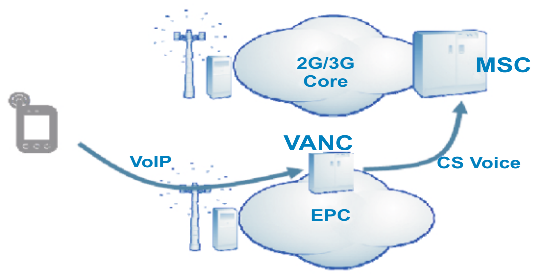


Figure 6 Working of VoLGA

NFC in cell Phones: The new Paradigm for an interactive world

WHAT IS NFC?

NFC, or near-field communication, is a variant of RFID, or radio frequency identification. It is an ultra short-range wireless technology that allows communication and data exchange between two devices held in tight proximity — about 4 cm apart.

HOW IS IT DIFFERENT FROM BLUETOOTH?

Bluetooth is also a short-range high frequency wireless technology but one that allows interaction between communication devices as much as 10 meters apart.

WHAT MAKES NFC SPECIAL?

NFC-enabled smartphones have the potential to replace credit cards. This is because NFC phones pack a smart chip — a complex 80-character code that is really hard to crack. Such a device can safely store confidential credit card details and be handy for purchases on the go. Frost & Sullivan predicts the technology will revolutionise e-commerce and drive over $150 billion worth of transactions by 2015, bulk of which is expected to be powered by NFC phones.

WHAT ELSE CAN THE TECHNOLOGY DO?

NFC can be deployed in ticketing services, rural banking, interactive and targeted advertising, healthcare, hospitality, libraries and pharmacies. In fact, an NFC phone could become the single-key to access to your car, home and office.

HOW DO NFC TRANSACTIONS WORK ?

Any device, a cellphone, a camera or a watch, can be equipped with an NFC 'initiator', which is simply an antenna that can store data. If the device is an NFC smartphone, the 'initiator' and 'target' (an NFC reader) need to be up close for data exchange to happen. The 'reader' is attached to a point-of-sale (PoS) terminal or cash-register in a retail store that accepts NFC payments. A simple wave of the phone can pay for a purchase. Alternatively, two NFC phones can be tapped lightly to exchange business cards.

WILL NFC BE A DRAIN ON BATTERY LIFE?

Geeks claim that in standby mode, a well-designed NFC solution does not consume any power. And since transactions happen in seconds, the power drain is not huge.

ARE NFC-ADOPTION LEVELS GROWING?

Globally, NFC adoption is picking up via smartphones. RIM, Nokia, Samsung and HTC have unveiled NFC smartphones. Apple iPhone5 is tipped to support NFC too. Google Wallet — a mobile payments technology that can be downloaded on some US mobile networks — is growing the NFC ecosystem. Payment trials have also begun in Australia, Singapore and China.

HAS NFC ARRIVED IN INDIA?

The technology is still in its infancy here. As of now, the Reserve Bank does not recognise NFC mobile payment transactions and PoS terminals accepting NFC payments don't exist. But NFC-enabled phones like BlackBerry's Touch Bold 9900 and Curve 9360, Samsung's Nexus S and Galaxy S II and Nokia's C7, 700, 701 and 600 are available. For NFC to take off, RBI has to frame norms and banks, carriers, credit-card companies, apps developers and PoS terminal makers have to team up. But awareness levels are growing and NFC is making some waves in entertainment. Shah Rukh Khan-starrer Ra.One was the first movie to be marketed by Nokia using NFC technology. Armed with an NFC phone, you can download the movie content by merely tapping the device on a NFC-tagged movie poster at a Nokia priority outlet or a partner multiplex.

Source:Economic Times

On the network side, VoLGA only requires software enhancements to the circuit to packet gateways which already exist for GAN. No modifications are required on the MSC or the LTE core and access network nodes. This enables a rapid development and market introduction, especially in multi-vendor MSC network environments. Furthermore, VoLGA enables the use of all other circuit switched services over LTE without any modifications in the network.

On the LTE side, the VANC (VoLGA Access Network Controller) based on 3GPP GANC (Generic Access Network Controller) connects to the Packet Data Network Gateway (P-GW) via the standard SGi interface. Both signaling and user data traffic (i.e. the voice packets) are transported over this interface. From an LTE core network point of view the VANC looks like any other IP based external node and IP packets exchanged between a wireless device and the VANC are transparently forwarded through the Evolved Packet Core (EPC) network. VANC is the network element required by VoLGA which connects the circuit switched MSC to the packet switched LTE Network.

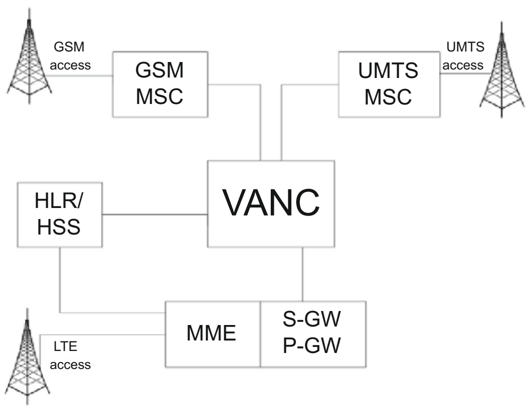
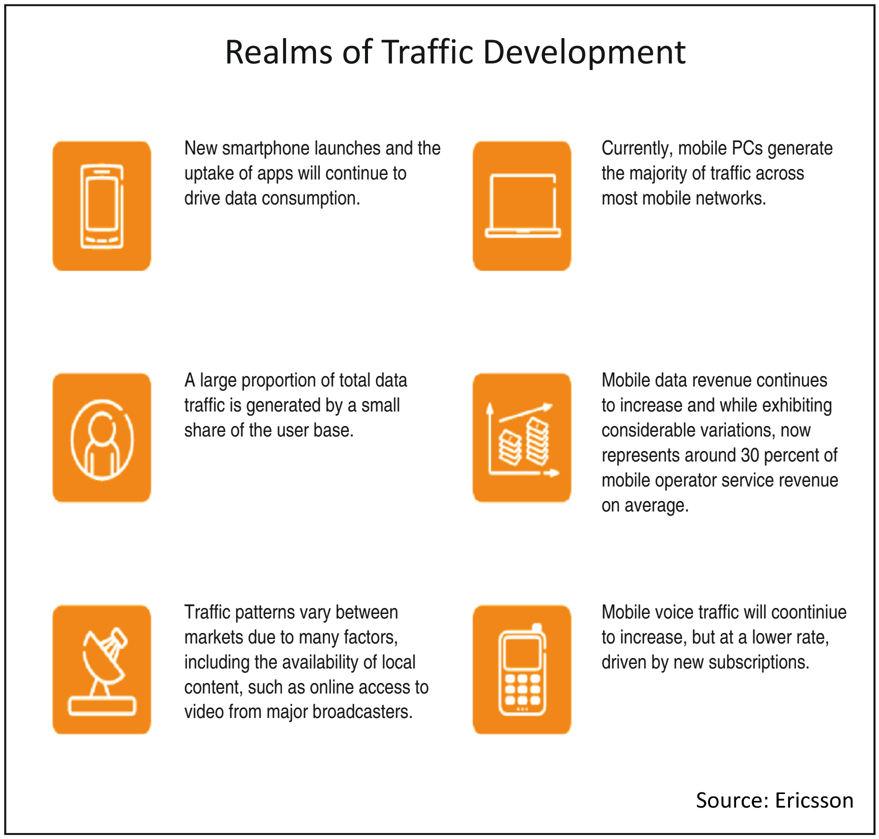


Figure 7 Basic VoLGA network setup

As no fallback to a legacy network is required, call setup times are not increased (as e.g for CS fallback) and the QoS is consistent with that of the 2G or 3G voice environment.

Although VoLGA offers a Voice over LTE solution with good QoS and low call setup times, it has not been accepted by 3GPP standardization bodies yet, which is a disadvantage compared to other voice over LTE solutions.



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