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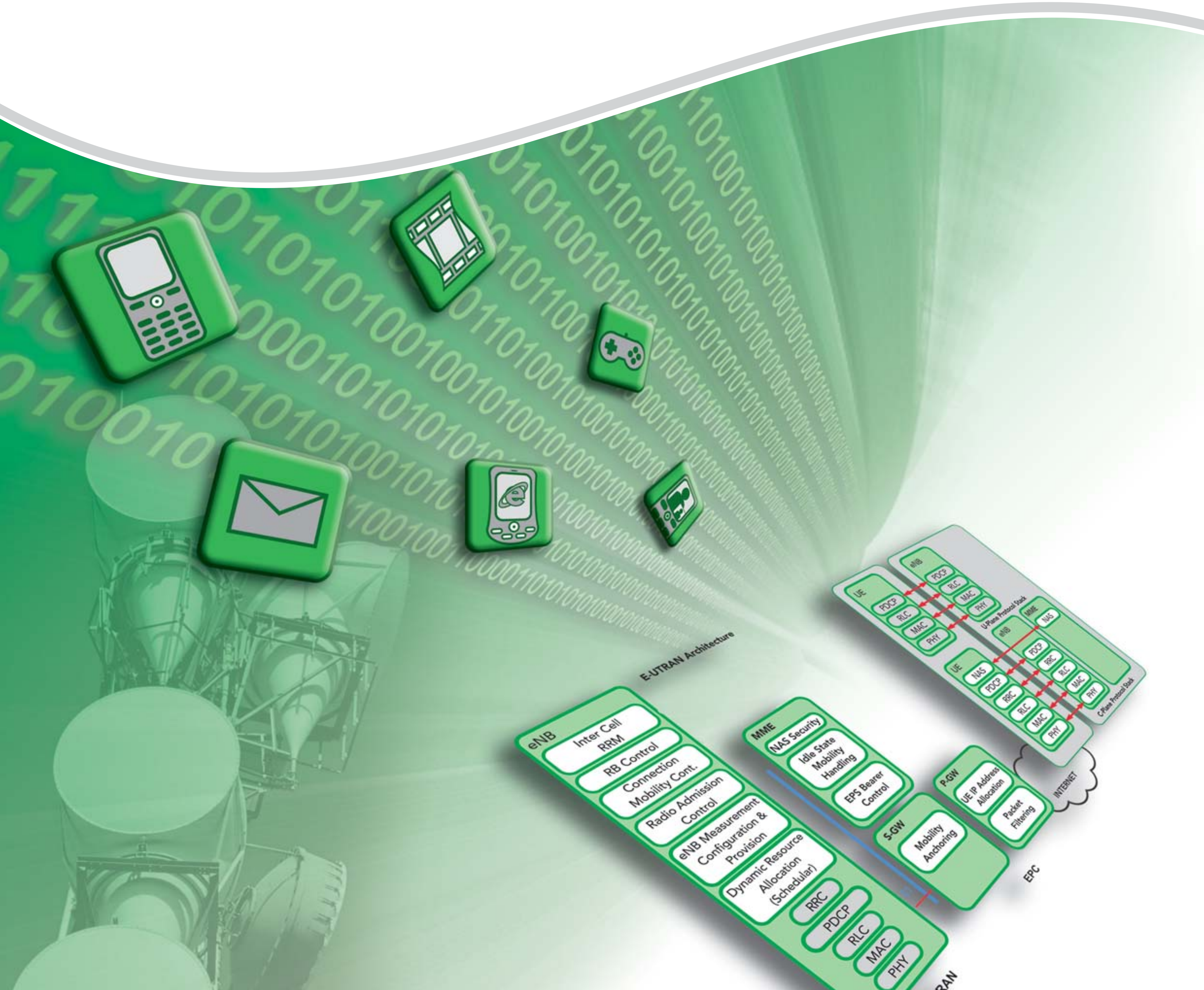
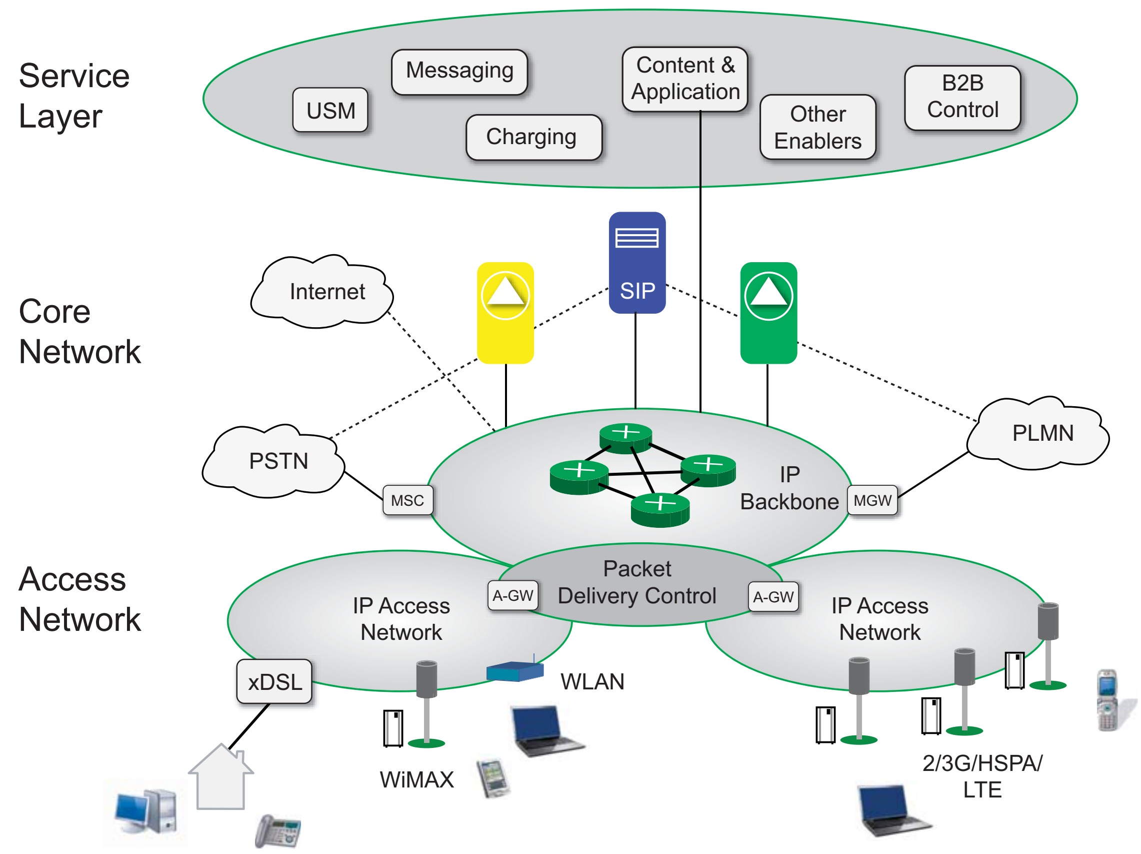
White Paper

Future technologies and testing for Fixed Mobile Convergence, SAE and LTE in cellular mobile communications.



Contents:

The Mobile Communications industry is currently developing standards and solutions for the next steps in the evolution of mobile networks. This introduction looks at the fundamental radio and network technologies being introduced in these steps, and how this is aligned into the future Fixed Mobile Convergence (FMC) of the Next Generation Networks.



Definition of Fixed-Mobile Convergence (FMC)

The following definition of FMC is based on the ETSI FMC ad hoc workgroup documents:

"Fixed and Mobile Convergence (FMC) is concerned with the provision of network and service capabilities, which are independent of the access technique. This does not necessarily imply the physical convergence of networks. It is concerned with the development of converged network capabilities and supporting standards. This set of standards may be used to offer a set of consistent services via fixed or mobile access to fixed or mobile, public or private networks.

An important feature of FMC is to allow users to access a consistent set of services from any fixed or mobile terminal, via any compatible access point. An important extension of this principle is related to roaming; users should be able to roam between different networks and be able to use the same consistent set of services through those visited networks as they have available in the home network. This feature is referred to as the Virtual Home Environment (VHE)."

FMC Motivations

The motivation behind FMC is to provide users with easy to use and desirable services, and to enable service providers to deliver this with cost effective networks. The user motivations are to enable more convenience with a required list of services as follows:

- Mobility of people and the need to communicate on the move is increased, and therefore the demand for mobile communications.
- Conventional fixed networks continue to serve the home or the office.
- Wide range of services within a uniform network and mobile connection is most important.
- Terminal mobility allows the customer the use of his (personal) terminal, e.g. his telephone at any place, at home, in the office or en route even abroad.
- Service mobility provides for the customer a common set of services independent of the access type and location. The services should have the "same look and feel" even in different networks.
- Personal mobility means reachability, in the sense that the customer is reachable with one number, his personal number, everywhere. He can define several reachability profiles (private, office) and he can change his profiles, especially the terminal where he wants his calls to arrive, from any terminal.

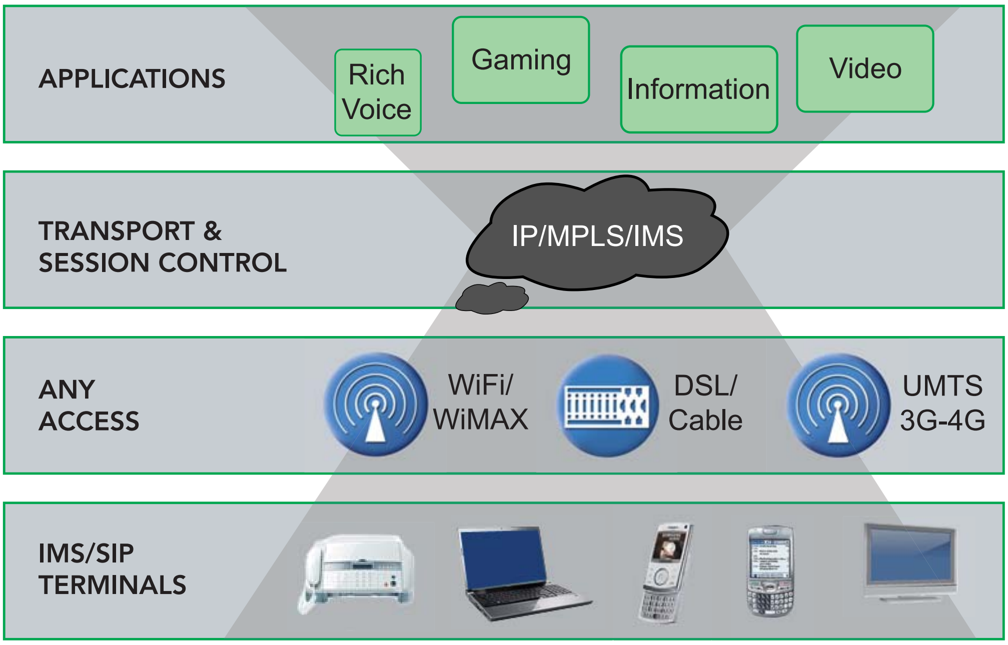
Fixed Mobile Convergence Operator Requirements

One of the key requirements to enable the vision of fixed mobile convergence is for convergence of the infra-structure and the O&M systems. Where an operator may today provide customer with multiple services (like fixed line voice and fixed line data to the home/office, mobile voice and data, multimedia TV and cable, interactive gaming and content etc), the operator must maintain separate management and control mechanisms for each service. A customer will still have a separate SIM card for the mobile, smart card for the cable/satellite, and usually separate billing mechanisms for each service.

To enable an operator to provide converged services, with a true 'triple play' offering, requires convergence of the core network, the data management, and customer care systems within the network. The evolution of the Next Generation Network, and the development of future cellular mobile networks, is towards providing the technical infrastructure and resources to enable this for service providers and network operators.

NGN Network Trend from technical point of view

The basic trend of Next Generation Network is towards an all IP network, to provide a simple method for extension of networks as growth demands increase, and to allow simple addition of new technologies to access the network. Traditionally, operators have built multiple networks to provide multiple services to customers (e.g. fixed telephone network, cable TV network, mobile network, xDSL data networks), but for the future we would aim for a single network that can provide all of these functions in a simple way.



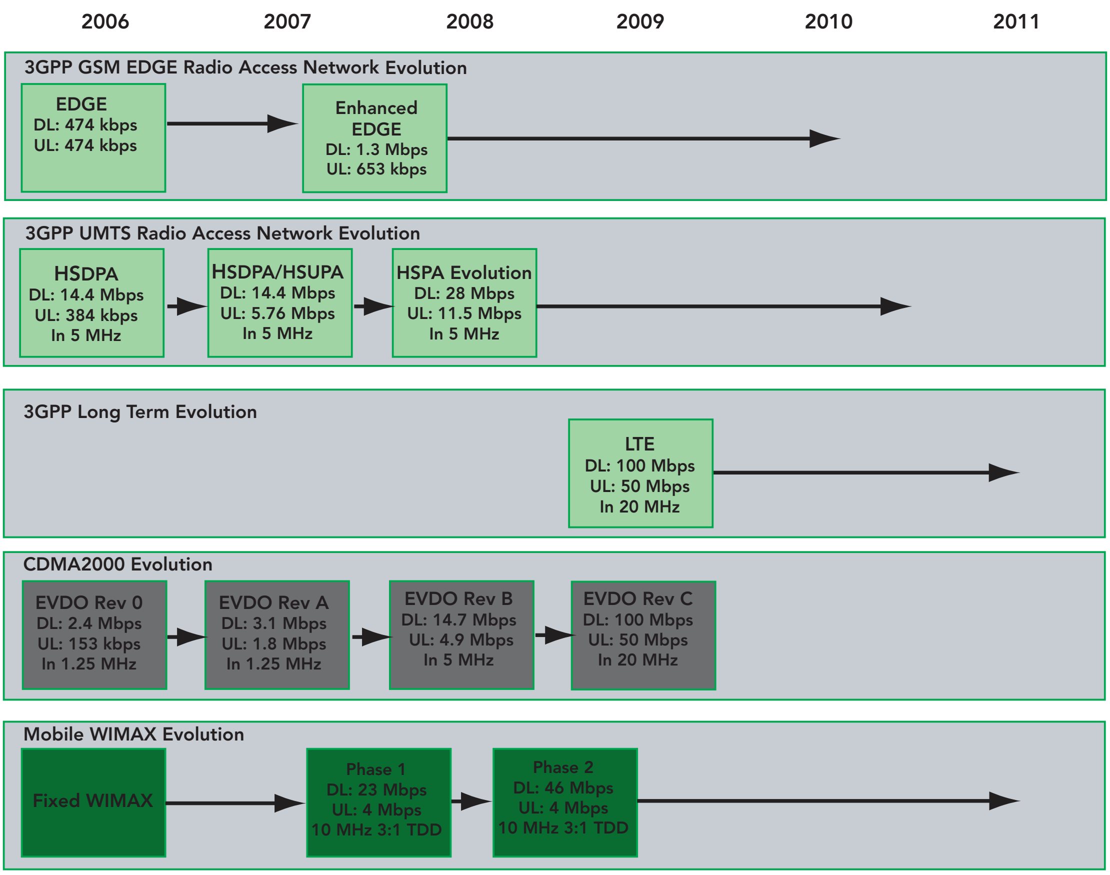
Technical Challenges in Converged Networks

So we see that the core network for NGN is an all IP network. The development of IP protocols has been supporting this requirement for some years now, and IPv6 has specifically included features that enable this vision. These developments include increased address ranges to provide sufficient addresses to support users with an individual unique IP address throughout the whole network, and to provide QoS support that is necessary for mobile networks and the wide variety of applications and services to be delivered in the network.

The mobile networks will be connected to the Core Network through the IP Multimedia Subsystem (IMS). The IMS will provide the necessary mobility and routing management required by a mobile network, and ensures that the core network sees the mobile network as another IP network. The core network will not need to manage mobility, authentication or security control as the user changes access technology in the mobile network. In today's network this is the case. For example, changing from WLAN access to GPRS data card on a laptop requires full connection, registration and authentication on each network and then manual control to switch from one to the other. Even when the mobile device supports both access technologies, the data flow can not be seamlessly handed over between the 2 access technologies with no user awareness of the change. The IMS allows this seamless handover between multiple access technologies, including management of billing, authentication and security access control. The IMS also uses Session Initiated Protocol (SIP) to allow fast connection between the mobile device and the core network. This is a key technology to enable the mobile network to feel like an IP network. Traditional wireless networks have an extended time period for initial set-up of a data session (typically 1-15 seconds) where a fixed network would provide this in milliseconds. The mobile networks are moving now to all IP so that they can be easily deployed in a mixed technology scenario, and enable a simple management and maintenance requirement without needing many different proprietary networks to be maintained.

The individual access technologies of the mobile networks are also evolving to provide higher data rates and improved spectral efficiency. There is also diversification in the deployment scenarios for each technology, so that an operator can mix multiple technologies into a single network to optimise network resources for local requirements. Examples of this are the use of WLAN hotspots for short range static users (e.g airport lounge), WiMAx for providing static wide area coverage, and HSPA for high speed mobile access.

In the chart below we can see evolution of new key technologies in the mobile access network that are developing to contribute towards this NGN vision.



Note: throughput rates are peak network rates. Radio Channel bandwidths indicated.
Dates refer to intial network deployment.

Wireless Communications Standards- Throughput Evolution

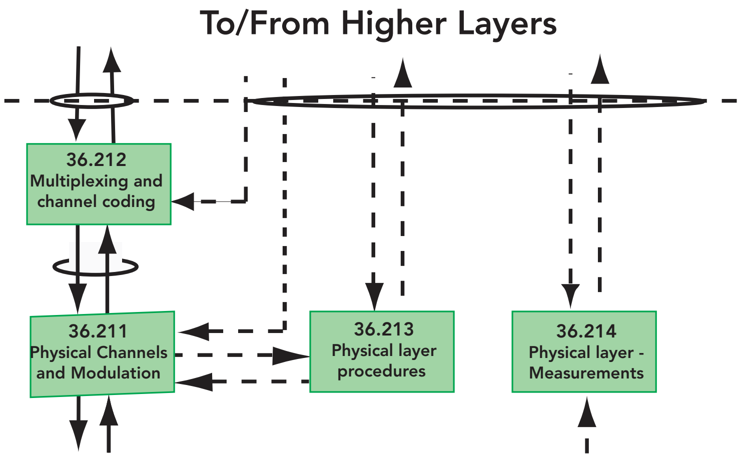
Having considered the overall NGN network requirements, and the top level technology and user requirements for this, we will now look at the evolution of the 3GPP network for mobile communications.

LTE/SAE Introduction

Release 7 of 3GPP includes study items to introduce MIMO and 64QAM as transmission technologies to increase data rate of the air interface, and IMS phase 2 introduces all IP network capability. This will reach the practical limits in data rates and capacity for the existing 3G networks based on 5 MHz WCDMA technology. Release 7 has been designed as an upgrade for existing HSPA networks, and is sometimes called "HSPA+", or "HSPA Evolution". Beyond here W-CDMA will still evolve in Release 8 and later, using multi-carrier technologies to increase data rates, and further improvements in signalling to increase data rates and reduce latency.

To move past the limitations of a 5 MHz W-CDMA system, 3GPP is now developing the standards for a new mobile network, and this is called the Long Term Evolution (LTE) and System Architecture Evolution (SAE) for next generation mobile networks. This is the next step in the continuous move to wider bandwidth and higher data rates. LTE and SAE are specified within 3GPP as part of the Release 8 version of specifications within the 36.xxx series of specifications. LTE will be a new 'evolved' radio interface and access network (E-UTRAN – Evolved Universal Terrestrial Radio Access Network), but will co-exist with WCDMA (UTRAN) that will also continue to evolve within 3GPP. The overall description for LTE/SAE is in TS 36.300, and the architecture description is in TS 36.401. E-UTRAN will also further evolve in 3GPP Release 9, and then into IMT-Advanced as Release 10 & Release 11 for data rates towards 1 GB/s.

The physical layer specification for LTE consists of a general document (TS 36.201), and four documents (TS 36.211 through 36.214). The relation between the physical layer specifications in the context of the higher layers is shown below.



- TS 36.201: Physical layer – general description
  The contents of the layer 1 documents (TS 36.200 series); Where to find information; general description of LTE layer 1.
- TS 36.211: Physical channels and modulation
  To establish the characteristics of the layer-1 physical channels, generation of physical layer signals and modulation. Definition of the uplink and downlink physical channels; The structure of the physical channels, frame format, physical resource elements, etc.
- TS 36.212: Multiplexing and channel coding
  To describe the transport channel and control channel data processing, including multiplexing, channel coding and interleaving.
- TS 36.213: Physical layer procedures
  To establish the characteristics of the physical layer procedures.
- TS 36.214: Physical layer – measurements
  To establish the characteristics of the physical layer measurements.

Purpose:

As we discussed above, the purpose of LTE/SAE is to support the NGN, and Fixed Mobile Convergence through the IP Multimedia Sub-system. For the user, this means to provide an always connected high speed user experience, to feel just like an ADSL home network, but in a mobile environment. For the operator/service provider, this means to provide an integrated network that is simple and cost effective to deploy, and allows integration to the core network for customer care, billing, and management of the network.

So the key challenges are:

- Data rates to true xDSL rates (e.g. 25-100 Mb/s given by VDSL2+ technology).
- Connection set-up time, must give an 'always connected' instant feel.
- Seamless integration of Internet applications, unaffected by carrier technology.
- A cost effective network infrastructure, and attractive user terminals.

As the future networks are integrated into a single IP network, so they offer different types of services across a single network, and there is a requirement to differentiate the types of services and the demands they place on the network. One of the key requirements is to be able to specify the Quality of Service (QoS) requirements for the different services. This allows the mobile network to be configured according to user/application requirements. The QoS will indicate to the network how to prioritise different data links to users across the network, and how to manage the capacity so that all users and applications are able to operate correctly. Typically, the QoS may be specified as a required or minimum data rate, and required error control and re-transmission procedures. In the table below we can see some examples of types of services, and the corresponding requirement s they have on the QoS and latency in the network.

Class of service	Bandwidth	Latency	QoS Requirement	Example
Conversational	Low-medium	Low	Guaranteed	VoIP/Video calling
Streaming	High	Low	Guaranteed	IPTV, multi-media streaming
Browsing	Low-medium	Normal	Best Effort	Web browser
Background	Medium	Normal	Minimal	Email synchronisation
Broadcast	High	Low	Guaranteed	Multi-cast

The Technologies:

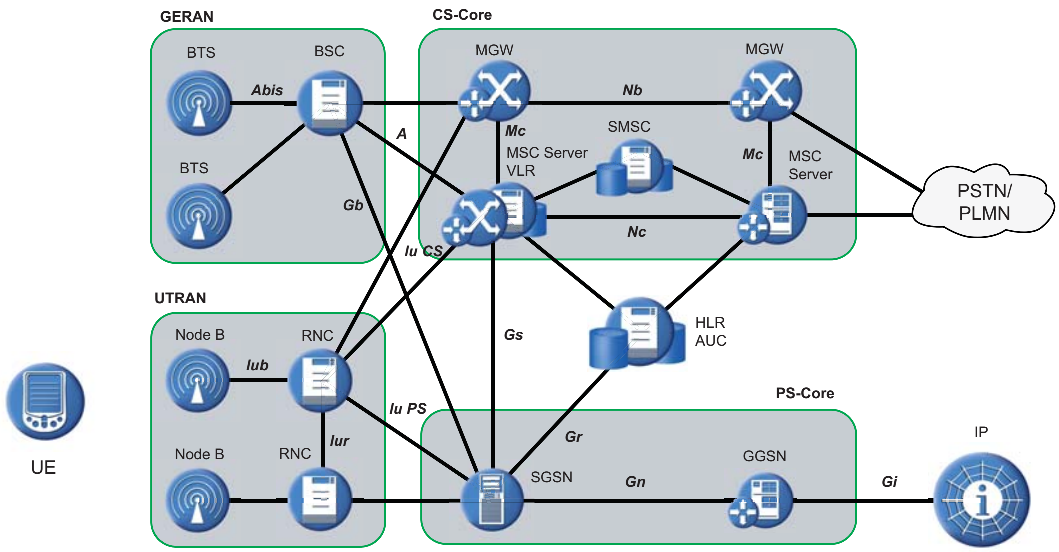
The two technologies we will consider are LTE (Long Term Evolution) and SAE (System Architecture Evolution). These two technologies address the future requirements of the radio access network (RAN) and core network (CN) respectively in the mobile network. These have now become known as the Evolved Universal Terrestrial Radio Access Network (E-UTRAN) and Evolved Packet Core (EPC).

First we will look at the SAE aspects, to understand the overall architecture of the new network and technology, and then we will look more closely at the LTE radio network.

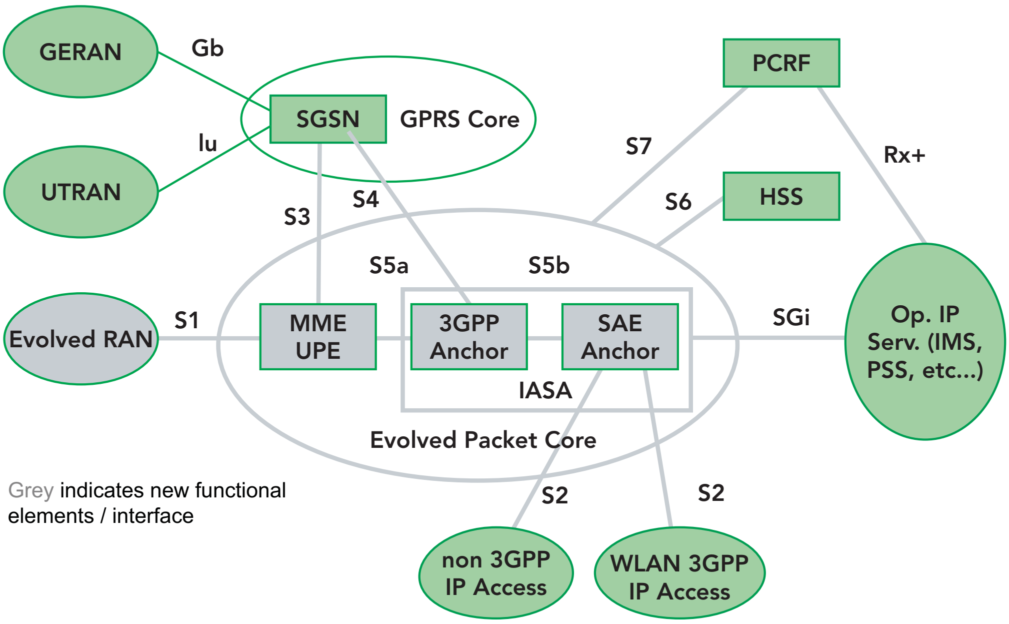
SAE Technology

SAE is the network architecture and design to simplify the network and provide seamless integration of the mobile network to other IP based communications networks. SAE uses a new evolved Node B (eNB) and Access Gateway (aGW) and removes the RNC and SGSN from the equivalent 3G network architecture, to make a simpler mobile network. This allows the network to be built as a "flat all-IP" based network architecture. SAE also includes entities to allow full inter-working with other related Wireless technologies (WCDMA, WiMAX, WLAN etc.). These entities can specifically manage and permit the non-3GPP technologies to interface directly into the network and be managed from within the same network.

As a reference, we can look at an existing 3G network architecture, shown below. First of all, we can notice that this is shown with only the packet network included, as the SAE will be only packet based. A full 3GPP network today includes a circuit switched network. This circuit switched element is an evolution of the original GSM voice network architecture from the 1990's. One objective of NGN was to move away from this old legacy network into a single all IP network using VoIP technology to provide voice services, rather than a separate circuit switched network.



This old packet network is based around the GGSN (gateway to external networks) and the SGSN (managing mobility and routing within the wireless network). There is no direct link from this network through to any other network that may be used as a complementary access technology. So, to link through to a WLAN or WiMAX network requires connection through either the public network (PDN) or through some proprietary IMS sub-system that an operator may implement for his own network. Either way, this does not provide a simple and extendable architecture that can meet the future needs of wireless communications.



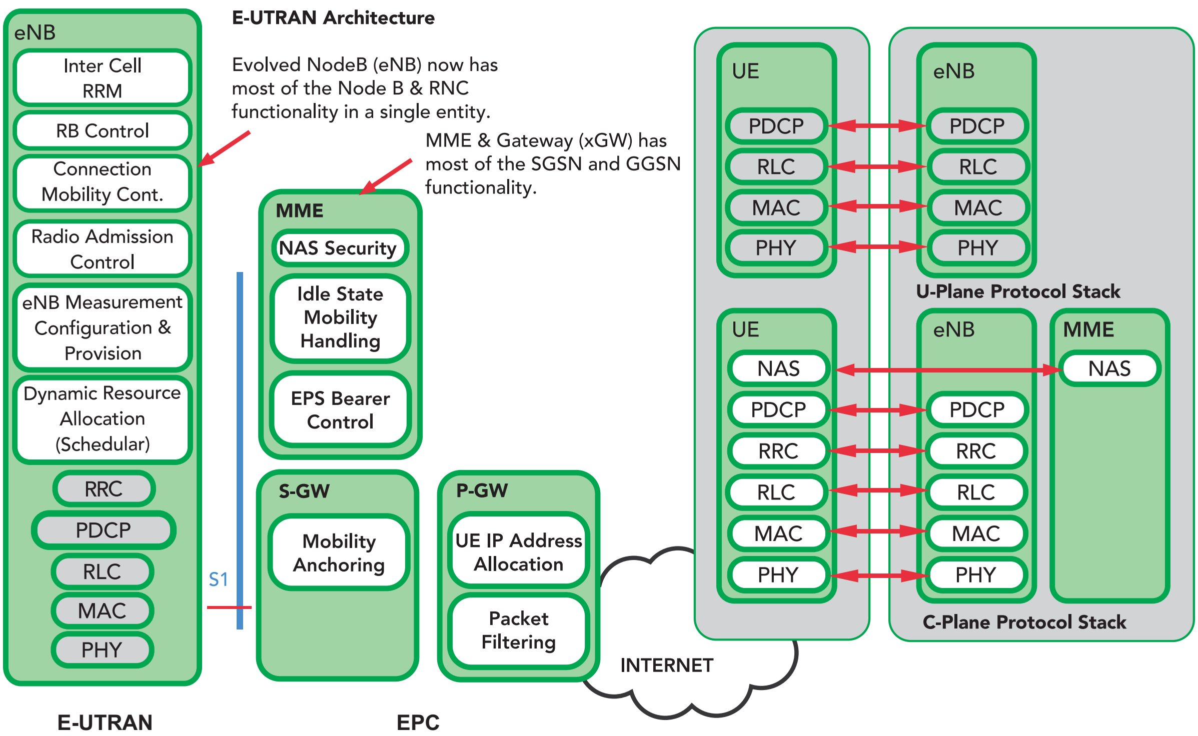
3GPP Anchor - The 3GPP Anchor is a functional entity that anchors the user plane for mobility between the 2G/3G access system and the LTE access system.

SAE Anchor - The SAE Anchor is a functional entity that anchors the user plane for mobility between 3GPP access systems and non-3GPP access systems.

Future Network Architecture - Generic

For the future SAE, we can see that it consists of an Evolved Packet Core (EPC), which is simplified when compared to existing 3GPP networks, and has specific functions built in that allow direct connection and extension to other wireless networks. The 'S2' interface allows operators to extend the network to other IP based access technologies whilst still managing the critical functions like mobility, hand-over, billing authentication and security within the mobile network. The EPC uses the 'S1' interface to connect to the wireless radio access network (LTE), and the 'S3' interface to connect data through to the SGSN to support handover to older 3GPP GPRS networks.

The EUTRAN network is broken down into two physical elements, the eNB and the aGW. This is considerably simpler than the previous 3G networks, with the equivalent to the RNC now being completely removed. Most of the flow control and data management functions of the RNC are now located in the eNB. The eNB is able to manage all transmission related issues at the transmit site, for faster re-transmission and link adaptation control. Previously these controls were passed through the network for the RNC to manage, and this would create additional round-trip delays. This allows for faster response time (e.g. for scheduling and re-transmissions) to improve latency and throughput of the network. The aGW manages all mobility and routing through the network, and also the link through to the authentication and billing databases.



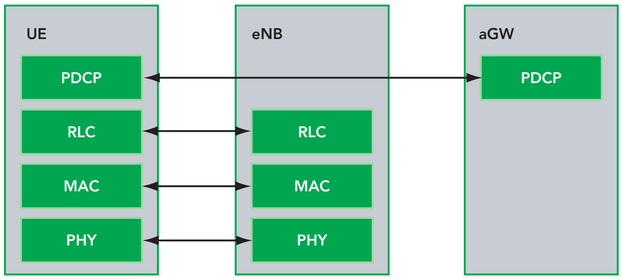
LTE Radio Access Network - Physical Elements

The diagram above shows the physical breakdown in of the network functionality into the eNB and the EPC (Evolved Packet Core). The EPC consists of a Gateway controller section that acts as a pipe to transfer user data between the eNB and the external network, and a Mobility Management section that manages mobility and routing of the data into the appropriate eNB. Here we can see the functions of the old NodeB and RNC now included into the eNB. On the right side of the diagram the basic protocol stack for user plan and control plan signalling is shown.

On the next diagram we can see the breakdown of the various elements in the protocol stack, showing the separation between eNodeB and MME, and the differences between User Plane and Control Plane signalling. We can see that in both case the full signalling up to PDCP layer takes place between UE and eNodeB. This is a key difference from previous networks, where the higher layer signalling was relayed back through the network to other network elements such as an RNC or MSC. This change means that the response to signalling is much faster as signals are not relayed through the network, and this contributes a significant amount to the reduced latency in LTE/SAE networks.

User Plane

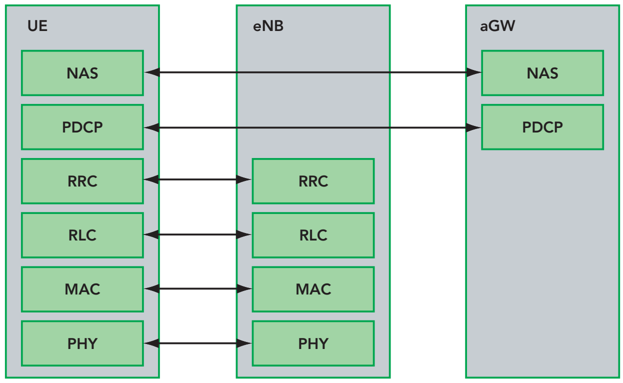
- RLC and MAC sublayers (terminated in the eNB on the network side) perform the functions such as:
  - Scheduling; ARQ; HARQ
- PDCP sublayer (terminated in aGW on the network side) performs for the U-plane the functions such as:
  - Header compression;
  - Integrity Protection (to be determined during WI phase)
  - Ciphering



U-plane Protocol Stack

C Plane

- RLC and MAC sublayers (terminated in eNB on the network side) perform the same functions as for the U-plane;
- RRC (terminated in eNB on the network side) performs functions such as:
  - Broadcast; Paging; RRC; connection management;
  - RB control; Mobility functions; UE measurement reporting and control.
- PDCP sublayer (terminated in aGW on the network side) performs for the C-plane functions such as:
  - Integrity Protection; Ciphering.
- NAS (terminated in aGW on the network side) performs among other things:
  - SAE bearer management; Authentication;
  - Idle mode mobility handling;
  - Paging origination in LTE_IDLE;
  - Security control for the signalling between aGW and UE, and for the U-plane.



C-plane Protocol Stack

New Radio Access Network - Protocol Elements

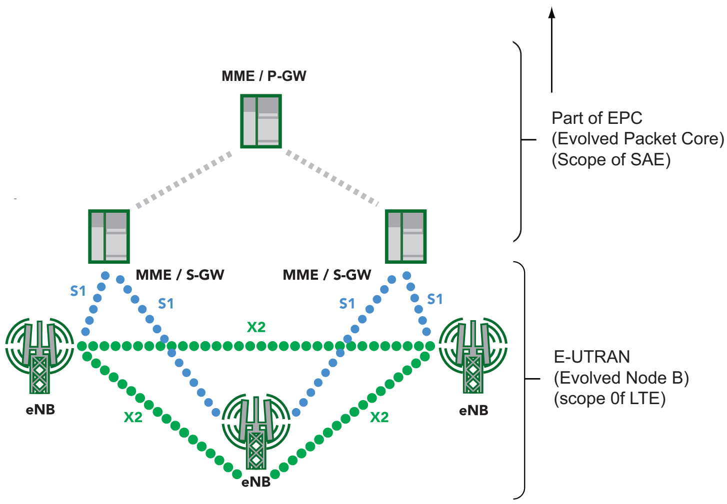
The logical breakdown of functions is shown in the next diagram. It is seen that the aGW can support multiple eNB's across multiple S1 interfaces, so the aGW control plane will be responsible for mobility in the network. When looking at the logical elements in the network, we see two key elements in the aGW, these are the User Pane Entity (UPE) and the Mobility Management Entity (MME).

The eNB hosts the following functions:
- Functions for radio resource management; radio bearer control, radio admission control, connection mobility control, dynamic allocation of resources to UEs in both uplink and downlink (scheduling);
- IP header compression and encryption;
- Routing of user plane data towards server gateway;
- Measurement and reporting for mobility and scheduling.

The MME hosts the following functions:
- NAS signalling; -NAS signalling security; AS security control;
- Inter CN node signalling for mobility between 3GPP access networks;
- Idle mode UE reachability

The Serving Gateway (S-GW) hosts the following functions:
- The local mobility anchor point for inter-eNB handover;
- Mobility anchoring for inter-3GPP mobility;
- E-UTRAN idle mode downlink packet buffering and initiation of network triggered service request procedure;
- Transport level packet marking in the uplink and the downlink

the PDN Gateway (P-GW) hosts the following functions:
- Per-user based packet filtering (by e.g. deep packet inspection)
- UE IP address allocation;
- UL and DL service level charging, gating and rate enforcement
- DL rate enforcement based on AMBR;



LTE Radio Access Networks - Logical Elements

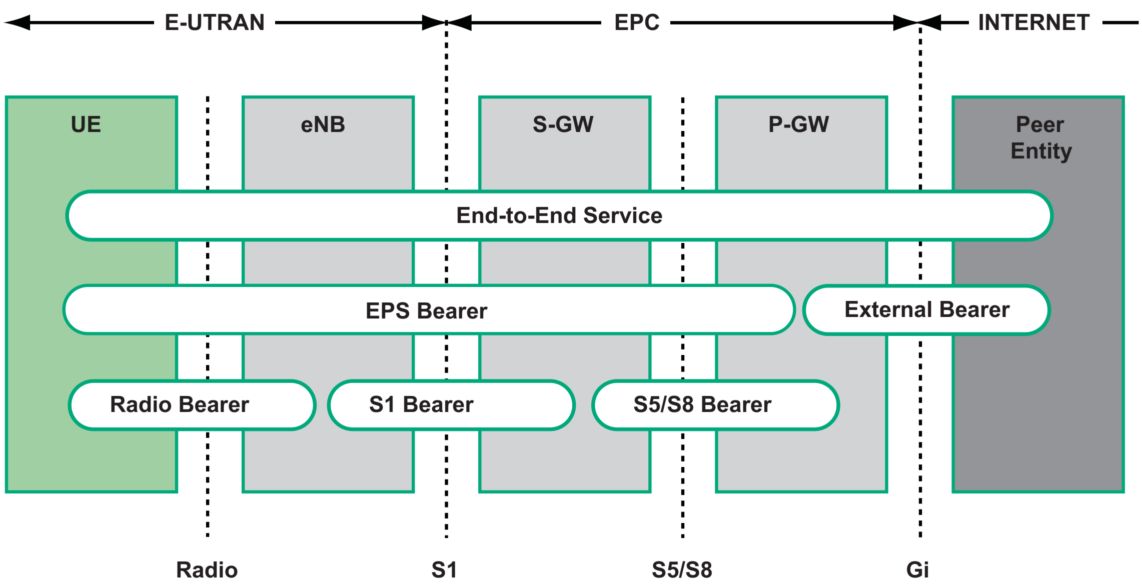
The MME is responsible for managing the eNB's and distributing paging messages to them. This allows for management of mobility in the network, through the correct distribution of paging messages to locate and provide data control information to the relevant users and the respective eNB's they are connected to. The UPE is responsible for the routing and delivery of the user data to/from the correct eNB's. This means that the user data IP headers and routing will be managed here, to ensure that data flows to the correct eNB and with the correct information for end user ID, QoS etc that are required by the eNB scheduling algorithms.

The UPE will also provide termination of the protocol stack for paging messages coming from a user through the eNB's. This is because paging messages are related to mobility and access requests within the mobile network, and are not related to data that is passed out of the network to an external application. So there must be correct termination of the protocol stack to permit the functions to work. As the mobility and admission control is managed from the aGW, the protocol stack for these functions is terminated here.

Bearer Service Architecture.

The end to end connectivity through the LTE/SAE network is made via the bearer service, which describes the 'top level' connectivity across the network.

- A radio bearer transports the packets of an EPS bearer between a UE and an eNB. There is a one-to-one mapping between an EPS bearer and a radio bearer.
- An S1 bearer transports the packets of an EPS bearer between eNodeB and Serving GW.
- An S5/S8 bearer transports the packets of an EPS bearer between a Serving GW and a PDN GW.
- A UE stores a mapping between an uplink packet filter and a radio bearer to create the binding between an SDF and a radio bearer in the uplink.
- A PDN GW stores a mapping between a downlink packet filter and an S5/S8a bearer to create the binding between an SDF and an S5/S8a bearer in the downlink.
- An eNB stores a one-to-one mapping between a radio bearer and an S1 to create the binding between a radio bearer and an S1 bearer in both the uplink and downlink.
- A Serving GW stores a one-to-one mapping between an S1 bearer and an S5/S8a bearer to create the binding between an S1 bearer and an S5/S8a bearer in both the uplink and downlink.



LTE Technology

Target Performance objectives for LTE.

When the project to define the evolution of 3G networks was started, the following targets were set by Network Operators as the performance design objectives. It was against these objectives that the different solutions were developed by various organisations and then proposed to 3GPP. The 3GPP then had a study to consider the proposals, evaluate the performance of each, and then make a recommendation for the way forward that would form the basis of LTE.

Peak data rate
- Instantaneous downlink peak data rate of 100 Mb/s within a 20 MHz downlink spectrum allocation (5 bps/Hz)
- Instantaneous uplink peak data rate of 50 Mb/s (2.5 bps/Hz) within a 20 MHz uplink spectrum allocation)

Control-plane latency
- Transition time of less than 100 ms from a camped state, such as Release 6 Idle Mode, to an active state such as Release 6 CELL_DCH
- Transition time of less than 50 ms between a dormant state such as Release 6 CELL_PCH and an active state such as Release 6 CELL_DCH

Control-plane capacity
- At least 200 users per cell should be supported in the active state for spectrum allocations up to 5 MHz

User-plane latency
- Less than 5 ms in unload condition (ie single user with single data stream) for small IP packet

User throughput
- Downlink: average user throughput per MHz, 3 to 4 times Release 6 HSDPA
- Uplink: average user throughput per MHz, 2 to 3 times Release 6 Enhanced Uplink

Spectrum efficiency
- Downlink: In a loaded network, target for spectrum efficiency (bits/sec/Hz/site), 3 to 4 times Release 6 HSDPA
- Uplink: In a loaded network, target for spectrum efficiency (bits/sec/Hz/site), 2 to 3 times Release 6 Enhanced Uplink

Mobility
- E-UTRAN should be optimized for low mobile speed from 0 to 15 km/h
- Higher mobile speed between 15 and 120 km/h should be supported with high performance
- Mobility across the cellular network shall be maintained at speeds from 120 km/h to 350 km/h (or even up to 500 km/h depending on the frequency band)

Coverage
- Throughput, spectrum efficiency and mobility targets above should be met for 5 km cells, and with a slight degradation for 30 km cells. Cells range up to 100 km should not be precluded.

Further Enhanced Multimedia Broadcast Multicast Service (MBMS)
- While reducing terminal complexity: same modulation, coding, multiple access approaches and UE bandwidth than for unicast operation.Provision of simultaneous dedicated voice and MBMS services to the user.
- Available for paired and unpaired spectrum arrangements.

Spectrum flexibility
- E-UTRA shall operate in spectrum allocations of different sizes, including 1.25 MHz, 1.6 MHz, 2.5 MHz, 5 MHz, 10 MHz, 15 MHz and 20 MHz in both the uplink and downlink. Operation in paired and unpaired spectrum shall be

supported .

- The system shall be able to support content delivery over an aggregation of resources including Radio Band Resources (as well as power, adaptive scheduling, etc) in the same and different bands, in both uplink and downlink and in both adjacent and non-adjacent channel arrangements. A “Radio Band Resource” is defined as all spectrum available to an operator.

Co-existence and Inter-working with 3GPP Radio Access Technology (RAT)

- Co-existence in the same geographical area and co-location with GERAN/UTRAN on adjacent channels.
- E-UTRAN terminals supporting also UTRAN and/or GERAN operation should be able to support measurement of, and handover from and to, both 3GPP UTRAN and 3GPP GERAN.
- The interruption time during a handover of real-time services between E-UTRAN and UTRAN (or GERAN) should be less than 300 msec.

Architecture and migration

- Single E-UTRAN architecture
- The E-UTRAN architecture shall be packet based, although provision should be made to support systems supporting real-time and conversational class traffic
- E-UTRAN architecture shall minimize the presence of "single points of failure"
- E-UTRAN architecture shall support an end-to-end QoS
- Backhaul communication protocols should be optimised

Radio Resource Management requirements

- Enhanced support for end to end QoS
- Efficient support for transmission of higher layers
- Support of load sharing and policy management across different Radio Access Technologies

Complexity

- Minimize the number of options
- No redundant mandatory features

So we can see that LTE refers to a new radio access technology to deliver higher data rates (50-100 MB/s), and fast connection times. The technology solution chosen by 3GPP uses OFDMA access technology, and MIMO technologies together with high rate (64QAM) modulation. LTE uses the same principles as HSPA (in existing Release 6 3GPP networks) for scheduling of shared channel data, HARQ, and fast link adaptation (AMC adaptive modulation and coding). This technology enables the network to dynamically optimise for highest cell performance according to operator demands (e.g. speed, capacity etc).

After evaluation of the different industry proposals, a recommendation was made to adopt an OFDMA based approach as part of a completely new air interface. The rationale for this was to chose ‘revolution’ rather than ‘evolution’ because in the long term this new air interface would offer the required data rates with the ability to implement in relatively low cost and power efficient hardware. It was felt that an ‘evolutionary’ approach based on further enhancement to WCDMA would be able to meet the technical requirements, but the technology demands to implement this may be unsuitable for mobile devices when considering power consumption, processing power etc. The OFDM based technology offers a simpler implementation of the required high speed data rates.

The performance of the selected technology has been modelled, and is predicted to meet the original requirements laid out in the LTE requirements specification.

The key features of the LTE air interface are:

Downlink

- OFDMA based access, with QPSK, 16QAM 64QAM modulation

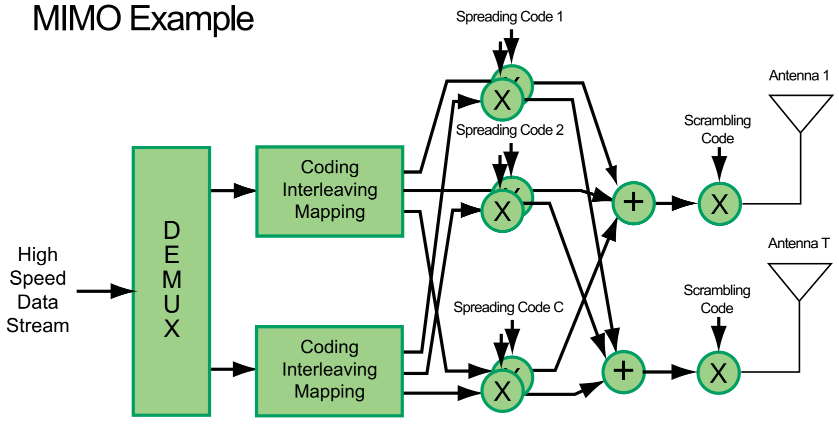
- Downlink multiplexing
- MIMO and transmit diversity
- MBM
- Scheduling, link adaptation, HARQ and measurements like in 3.5G

Uplink
- Single Carrier FDMA access, with BPSK, QPSK, 8PSK and 16QAM modulation
- Transmit diversity
- Scheduling, link adaptation, HARQ and measurements like in 3.5G
- Random Access Procedures

Multiple Input Multiple Output (MIMO)
- Employs multiple antennas at both the base station transmitter and terminal receiver
- If multiple antennas are available at the transmitter and receiver, the peak throughput can be increased using a technique known as code re-use.
    - Each channelisation/scrambling code pair allocated for PDSCH transmission can modulate up to M distinct data streams, where M is the number of transmit antennas.
- In principle, the peak throughput with code re-use is M times the rate achievable with a single transmit antenna.
- Compared to the single antenna transmission scheme with a larger modulation constellation to achieve the same rate, the code re-use technique may have a smaller required Eb/No, resulting in overall improved system performance.

OFDMA is used to provide higher data rates than used in 3G through the use of a wider transmission bandwidth. The 5 MHz channel is limiting for WCDMA data rates, and use of a wider RF band (20 MHz) leads to group delay problems that limit data rate. So, OFDM breaks down the 20 MHz band into many narrow bands (sub-channels) that do not suffer from the same limitation. Each sub-channel is modulated at an optimum data rate and then the bands are combined to give total data throughput. Algorithms select the suitable sub-channels according to the RF environment, interference, loading, quality of each channel etc. These are then re-allocated on a burst by burst level (each sub-frame, or 1 ms). The OFDM frequencies in LTE have been defined with a carrier spacing of 15 kHz. Each 'resource block' (that represents an allocation of radio resource to be used for transmission) is a group of 12 adjacent sub-carriers (therefore 180 kHz) and a slot length of 0.5 ms.

MIMO Example



We consider an example with 2 transmit antennas (A and B). The 3 options for transmission correspond to only antenna A, and only antenna B, and both A and B; options 1, 2 and 3.

When both antennas are on, the transmit power from each is reduced by a half so the total power is the same as the cases with a single antenna on. For each option, the SINR is calculated. The SINRs are mapped to achievable rates and their corresponding modulation and channel coding schemes, possibly based on conventional HSDPA rate mappings.

In the example, option 1 achieves a rate of 2.332 Mbps by using 16QAM modulation with a rate 0.486 coding over 5 spreading codes. Option 2 achieves a slightly higher rate because its SINR is higher. However, by transmitting simultaneously over both antennas, the total rate of 3.074 Mbps is the highest.

Option	TX antenna power		SINR (dB)		Modulation		Coding Rate		Number of Codes		Data Rate (Mbps)		
	A	B	A	B	A	B	A	B	A	B	A	B	Total
1	P	0	14	-inf	16QAM	N/A	0.486	N/A	5	N/A	2.332	0	2.332
2	0	P	-inf	15dB	N/A	16QAM	N/A	0.551	N/A	5	0	2.644	2.644
3	P/2	P/2	10.2	11.9	QPSK	16QAM	0.691	0.371	4	5	1.292	1.782	3.074

MIMO is an abbreviation of "Multiple Input Multiple Output". This is an antenna technology together with signal processing that can increase capacity in a radio link. In LTE, the user data is separated into 2 data streams, and these are then fed to 2 separate TX antennas, and received by 2 separate RX antenna. Thus the data is sent over 2 separate RF paths. The algorithm used to split and then recombine the paths allows the system to make use of the independence of these 2 paths (not the same RF losses and interference on both) to get extra data throughput better than just sending the same data on 2 paths. This is done by separating the data sets in both space and time. The received signals are then processed to be able to remove the effects of signal interference on each, and thus creating 2 separate signal paths that occupy the same RF bandwidth at the same time. This will then give a theoretical doubling of achievable peak data rates and throughput in perfect conditions where each RF path is completely isolated and separate from the other path.

LTE Protocols and Signalling

When 3GPP started to develop the radio interface protocols of the Evolved UTRAN the following initial assumptions were made:

- Simplification of the protocol architecture and the actual protocols
- No dedicated channels, and hence a simplified MAC layer (without MAC-d entity)
- Avoiding similar functions between Radio and Core network.

LTE Physical Layer Channel Structure

The E-UTRAN has been designed as part of an "all IP network", see the section on SAE for more details. But, some basic consequences from this are that there are no longer any circuit switched elements in the network, everything is now packet based. Further, the use of shared and broadcast channels that are introduced in earlier versions of 3GPP (e.g. HSDPA, HSUPA, MBMS) are re-used in LTE. In fact, the design is based fully on shared and broadcast channels, and there are no longer any dedicated channels to carry data to specific users. This is to increase efficiency of the air interface, as the network can control the use of the air interface resources according to the real time demand from each user, and no longer has to allocate fixed levels of resource to each user independent of real time data requirements.

The protocol stack for LTE is based on a standard SS7 signalling model, as used with 3GPP W-CDMA networks today. In this model, in layer 3 the Non Access Stratum (NAS) is connected to the logical channels. These logical channels provide the services and functions required by the higher layers (NAS) to deliver the applications and services. The logical channels are then mapped onto transport channels in Layer 2, using the RRC elements. These provide control and management for the flow of the data, such as re-transmission, error control, and prioritisation. User data is managed in Layer 2 by the Packet Data Convergence Protocol (PDCP) entity. The air interface and physical layer connection is then controlled and managed by the Layer 1, using RLC and MAC entities and the PHY layer. Here the transport channels are mapped into the physical channels that are transmitted over the air.

LTE radio channels are separated into 2 types, physical channels and physical signals. A physical channel corresponds to a set of resource elements carrying information originating from higher layers (NAS) and is the interface defined between 36.212 and 36.211. A physical signal corresponds to a set of resource elements used only by the physical layer (PHY) but does not carry information originating from higher layers.

Based on the above architecture, the downlink consists of 5 physical channels and 2 physical signals:

Downlink Physical Channels

1. Physical broadcast channel (PBCH)

   The coded BCH transport block is mapped to four subframes within a 40 ms interval, 40 ms timing is blindly detected, i.e. there is no explicit signalling indicating 40 ms timing.

Each subframe is assumed to be self-decodable, i.e the BCH can be decoded from a single reception, assuming sufficiently good channel conditions.

2. Physical control format indicator channel (PCFICH)
   Informs the UE about the number of OFDM symbols used for the PDCCHs;Transmitted in every subframe.

3. Physical downlink control channel (PDCCH)
   Informs the UE about the resource allocation, and hybrid-ARQ information related to DL-SCH, and PCH. Carries the uplink scheduling grant.

4. Physical downlink shared channel (PDSCH)
   Carries the DL-SCH. This contains the actual user data.

5. Physical multicast channel (PMCH)
   Carries the MCH. This is used to broadcast services on MBMS.

Downlink Physical signals

1. Reference signal
2. Synchronization signal

The corresponding uplink consists of 4 physical channels and 2 physical signals:

Uplink Physical Channels

1. Physical uplink control channel (PUCCH)
   Carries ACK/NAKs in response to downlink transmission. Carries CQI reports.

2. Physical uplink shared channel (PUSCH)
   Carries the UL-SCH. This contains the user data.

3. Physical Hybrid ARQ Indicator Channel (PHICH)
   Carriers ACK/NAKs in response to uplink transmissions.

4. Physical random access channel (PRACH)
   Carries the random access preamble.

Uplink Physical Signals

1. Demodulation reference signal, associated with transmission of PUSCH or PUCCH.
2. Sounding reference signal, not associated with transmission of PUSCH or PUCCH.

So we can see that both the uplink and downlink are composed of a shared channel (to carry the data) together with its associated control channel. In addition there is a downlink common control channel that provides non user data services (broadcast of cell information, access control etc…).

Within the LTE protocol stack, the physical layer channels (described above) are mapped through to the higher layers via the functions of the MAC and RLC layers of the protocol stack. This is shown below for both the user plane and control plane data. Here we can see the simplification of the network due to the SAE architecture discussed previously. The eNB is responsible for managing the air interface and flow control of the data, and the aGW is responsible for the higher layer control of user data within PDCP and NAS services.

LTE transport channel structure

Having now defined the physical layer channels, and the requirements of the higher layers in the protocol stack, we can now take a look at the transport channels types in EUTRAN, which are defined as follows:

Downlink transport channel types are:

- Broadcast Channel (BCH) characterised by:
  - fixed, pre-defined transport format;
  - requirement to be broadcast in the entire coverage area of the cell.

- Downlink Shared Channel (DL-SCH) characterised by:
  - support for HARQ;
  - support for dynamic link adaptation by varying the modulation, coding and transmit power;
  - possibility to be broadcast in the entire cell;
  - possibility to use beamforming;
  - support for both dynamic and semi-static resource allocation;
  - support for UE discontinuous reception (DRX) to enable UE power saving;
  - support for MBMS transmission (FFS).

NOTE: the possibility to use slow power control depends on the physical layer.

- Paging Channel (PCH) characterised by:
  - support for UE discontinuous reception (DRX) to enable UE power saving (DRX cycle is indicated by the network to the UE);
  - requirement to be broadcast in the entire coverage area of the cell;
  - mapped to physical resources which can be used dynamically also for traffic/other control channels.

- Multicast Channel (MCH) characterised by:
  - requirement to be broadcast in the entire coverage area of the cell;
  - support for SFN combining of MBMS transmission on multiple cells;
  - support for semi-static resource allocation e.g. with a time frame of a long cyclic prefix.

Uplink transport channel types are:

- Uplink Shared Channel (UL-SCH) characterised by:
  - possibility to use beamforming; (likely no impact on specifications)
  - support for dynamic link adaptation by varying the transmit power and potentially modulation and coding;
  - support for HARQ;
  - support for both dynamic and semi-static resource allocation.

NOTE: the possibility to use uplink synchronisation and timing advance depend on the physical layer.

- Random Access Channel(s) (RACH) characterised by:
  - limited control information; collision risk;

LTE Logical channel structure

To complete the channel mapping, the transport channels are mapped onto the logical channels in the protocol stack. These logical channels then provide the functionality to the higher layers in the protocol stack, and they are specified in terms of the higher layer services which they support. Each logical channel type is defined by what type of information is transferred. In general, the logical channels are split into two groups:

1. Control Channels (for the transfer of control plane information);
2. Traffic Channels (for the transfer of user plane information).

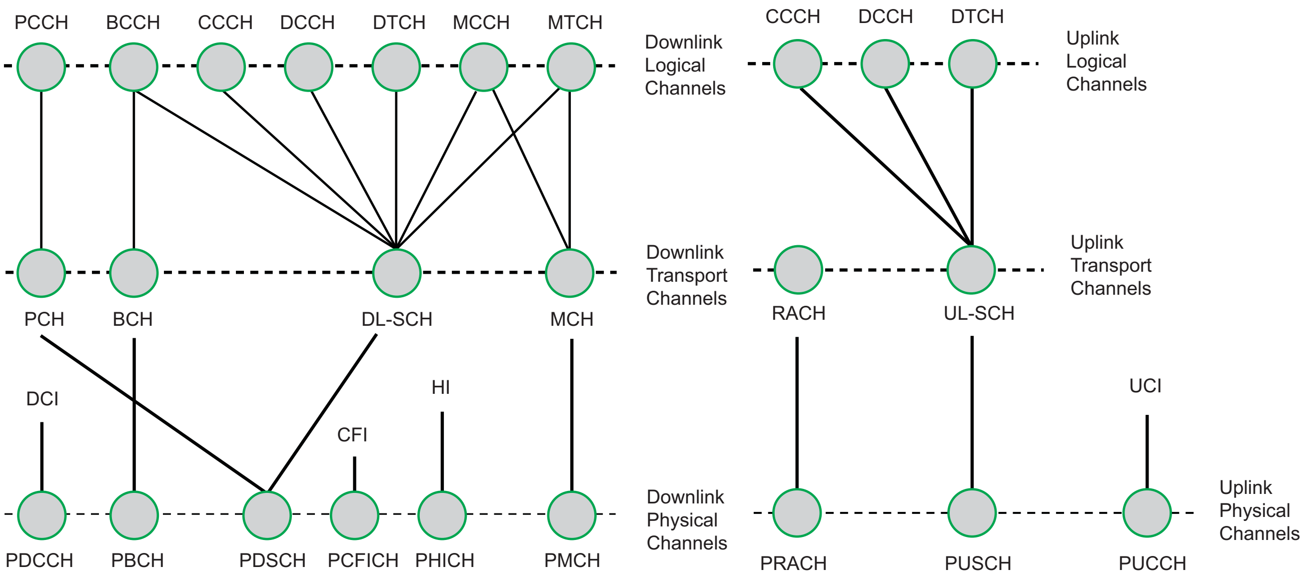
Control channels are used for transfer of control plane information only. The control channels are:

- Broadcast Control Channel (BCCH)
  A downlink channel for broadcasting system control information.
- Paging Control Channel (PCCH)
  A downlink channel that transfers paging information. This channel is used when the network does not know the location cell of the UE.
- Common Control Channel (CCCH)
- Multicast Control Channel (MCCH)
  A point-to-multipoint downlink channel used for transmitting MBMS control information from the network to the UE, for one or several MTCHs. This channel is only used by UEs that receive MBMS.
- Dedicated Control Channel (DCCH)
  A point-to-point bi-directional channel that transmits dedicated control information between a UE and the network. Used by UEs when they have an RRC connection.

Traffic channels are used for the transfer of user plane information only. The traffic channels are:

- Dedicated Traffic Channel (DTCH)
  A Dedicated Traffic Channel (DTCH) is a point-to-point channel, dedicated to one UE, for the transfer of user information. A DTCH can exist in both uplink and downlink.
- Multicast Traffic Channel (MTCH)
  A point-to-multipoint downlink channel for transmitting traffic data from the network to the UE. This channel is only used by UEs that receive MBMS.

The mapping from logical channels to transport channels to physical channels is shown below:



LTE Channel Mapping

MAC Scheduler

All user data transmissions in LTE are made by pre-scheduled transmissions of specific packets of data, there are no continuous 'circuit' connections as with previous cellular technologies, this is reflecting the 'mobile broadband' and 'all IP' nature of LTE. All of the scheduling for transmissions, both downlink and uplink, are made by the MAC scheduler in the eNodeB. This entity will decide the instantaneous data rate and capacity given to each user in the network. Resources are scheduled based on reported CQI and other factors. The algorithm for this is not specified by 3GPP, and so different eNodeB manufacturers and network operators can develop and tune the algorithms to give different performance or behaviours in their network.

The scheduling assignments are sent on the DCI/DPCCH channels, and are DL-SCH/PDSCH scheduling assignments for the UE to receive data, and UL-SCH/PUSCH scheduling assignments (UL grants) for the UE to transmit data. The UE can not set the uplink scheduling, as this must be co-ordinated across all UE's in the cell, and so this is done in the eNodeB. The UE can request UL-SCH resources by one of two methods, and then wait for the eNodeB to respond with a scheduling assignment:

- Explicitly by sending an SR (Scheduling Request) on UCI/PUCCH
- Implicitly by sending a BSR (Buffer Status Report) on UL-SCH/PUSCH

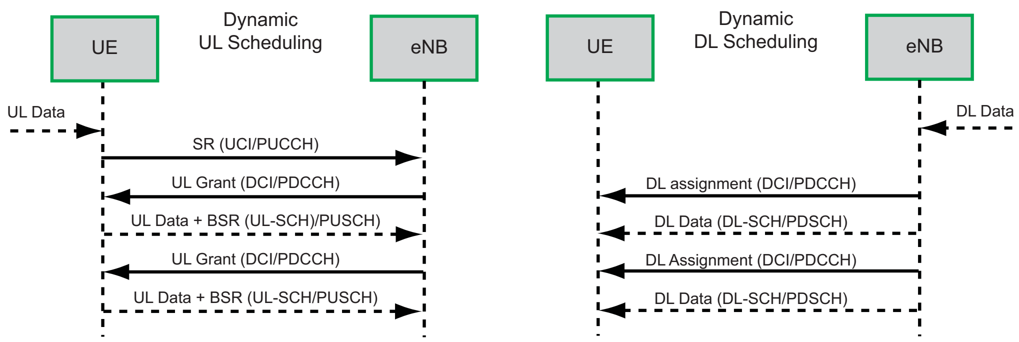
The MAC has 2 scheduling modes:

- Dynamic (based on dynamic requests and assignments)
- Semi-persistent (based on static assignments, for e.g. voice traffic)

The semi-persistent scheduling is used for applications where the network expects to have a continuous need to the same scheduling on a repetitive basis. An example is for voice calls, where the voice is coded and put into packets by the CODEC at a fixed rate, and then the packets need to be transmitted at a fixed rate/interval during the entire voice call. With this mode, the network can have a persistent schedule for the UE to receive each voice packet without the need to signal the UE in advance for each packet. This reduces the amount of control signalling required and hence makes more radio resource available for user data.

Inside the scheduler, the DRX sequence (Discontinuous Rx) has to be taken into account. This is the process where the UE can 'switch off' it's receiver for fixed time periods (to save power and increase stand-by time). The scheduler needs to know if the UE is likely to be 'asleep' in DRX mode, and if so then it needs to signal the UE to 'wake up' before sending it resource allocations.

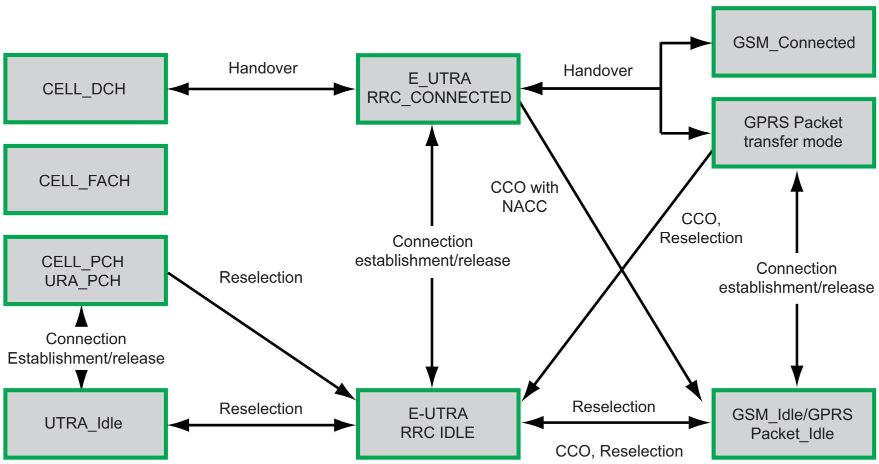
The process for dynamic scheduling is shown below:



MAC Dynamic Scheduling

Latency Improvements Within LTE

As part of the LTE protocol description, the RRC States were restricted to "RRC Idle" and "RRC Connected" States. They are depicted below, in conjunction with the possible legacy UTRAN RRC. The purpose of this is to simplify the protocol structure and number of possible states, and hence state transitions. This will enable faster response times in the network to reduce latency for the user. In an existing 3GPP WCDMA network there are more possible states and state transitions, and in addition it takes some considerable time to move from one state to another. This leads to excessive delays when trying to set up data links or re-configure the radio resources to be more efficient, to change the local loading in the cell, or change to the demand or a particular user.

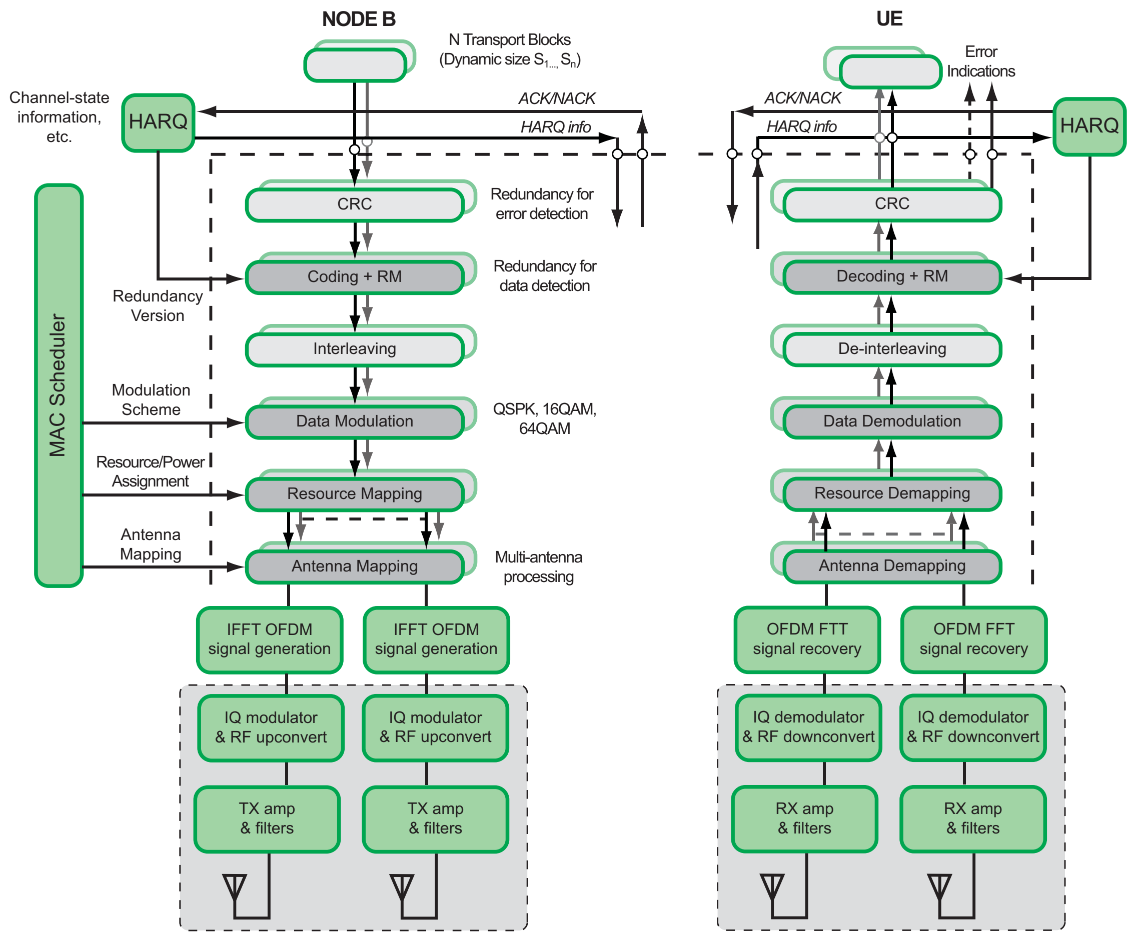


RRC States and Inter RAT Mobility Procedures

In addition, the Transmission Time Interval (TTI) of 1ms was agreed (to reduce signalling overhead and improve efficiency). The TTI is the minimum period of time in which a transmission or re-transmission can take place. Having a short TTI means that when messages are received, a reply can be scheduled much faster (the next available transmission slot will be sooner), or a re-transmission of a failed message can take place much sooner.

LTE Typical Implementations

A Typical implementation of the downlink is shown below, giving the various coding and processing stages to go from the logical channel (user/data stream) through to an air interface transmission and back through the receiver to recover the logical channel. The example shows a typical 'downlink' with 2 channels of MIMO.



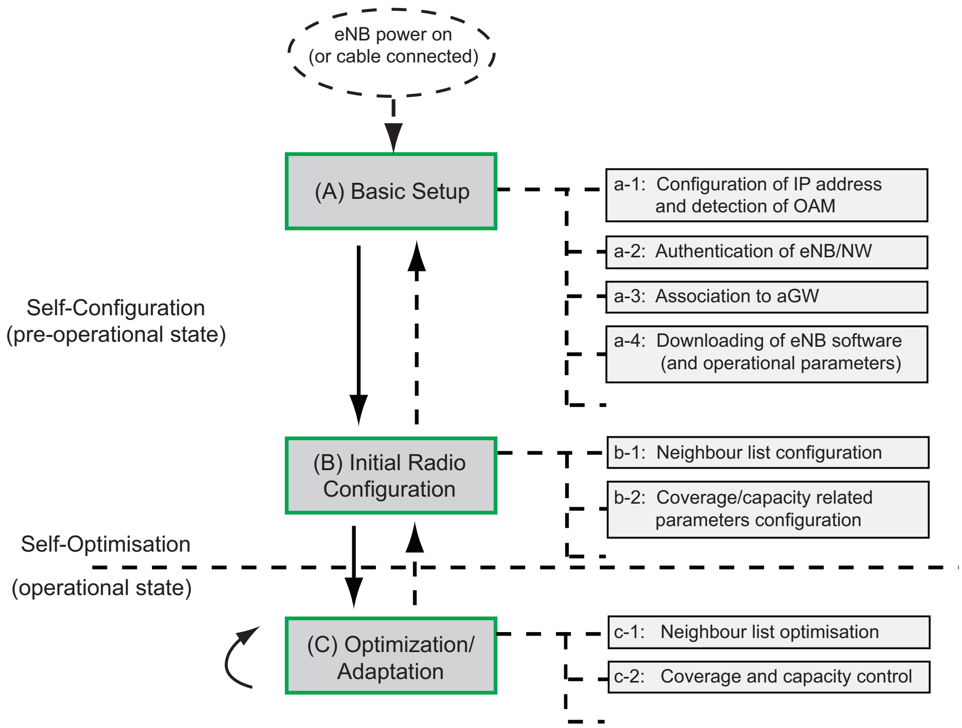
System Architecture Functional Diagram, Downlink Transmission and Reception

Self Optimising Networks (SON)

Another new concept being introduced for SAE networks is the Self-Optimising Network (or Self Organising Network). The purpose of this is to reduce the complexity of deploying new nodes into the network. This can be for new micro/macro cells in busy areas, for pico cells being installed into local hotspots such as shopping centres or airports, or even femto cells being installed for coverage within a single home.

Traditionally, when a new node is configured into the network there are a number of planning and measurements that must be made by experts in the field, and then parameter setting and network tuning will take place using other experts who are managing the network. This is all to do with setting network power levels, neighbour cell lists for handover measurements, and configuring the node correctly into all other network databases.

The concept for SON is to allow this to take place automatically using software in the eNodeB, so all necessary measurements are made automatically and reported to the network. In addition, the network should be able to automatically transfer this information across to other network elements that will use the information for configuration/optimisation purposes. This concept includes several different functions from eNB activation to radio parameter tuning. The chart below shows a basic framework for all self-configuration /self-optimization functions.



Self-configuration process is defined as the process where newly deployed nodes are configured by automatic installation procedures to get the necessary basic configuration for system operation.

This process works in pre-operational state. Pre-operational state is the state from when the eNB is powered up and has backbone connectivity until the RF transmitter is switched on.

The typical functions handled in the pre-operational state covered by the Self Configuration process are:

- Basic Setup
- Initial Radio Configuration
- Procedures to obtain the necessary interface configuration;
- Automatic registration of nodes in the system provided by the network;

Self-optimization process is defined as the process where UE & eNB measurements and performance measurements are used to auto-tune the network. This process works in operational state where the RF interface is additionally switched on.

The typical functions handled in the operational state covered by the Self Optimization process are:

- Optimization / Adaptation of network settings.
- The distribution of data and measurements over interfaces;
- Communication with functions/entities/nodes in charge of data aggregation for optimization purpose;
- Managing links with O&M functions and O&M interfaces relevant to the self optimization process.

To enable SON it is also necessary that the UE shall support measurements and procedures which can be used for self-configuration and self-optimisation of the E-UTRAN system.

To reduce the impact of SON on the UE (cost, complexity, battery life etc) measurements and reports used for the normal system operation should be used as input for the self-optimisation process as far as possible. For SON specific measurements required by the network, the network is able to configure the measurements and the reporting for self-optimisation support by RRC signalling messages sent to the UE.

Impact on Users of the Technology:

Consumers

The key impact of LTE/SAE and the FMC vision for users of mobile communications will be the feeling of 'always connected' with the ability to have 'what I want, when I want, and where I want' without having to care about how to access the services. The range of services offered to a consumer should become independent of the access technology, so that music download, instant messaging, voice and video calling all become available in the same format wherever you are.

The FMC will enable users to have a single contact number (IP address) that can be taken anywhere in the network, and the user can set profiles according to where they are and what they are doing. So a mobile phone should automatically be able to connect to a home network, an office network, a public network or a local hotspot without the user making any changes. The phone will then be able to configure itself into 'office mode' to provide office based services when in the office, and then change to home services when the user returns home.

By integrating the mobile network to the fixed network through the IMS sub-system, the look and feel of the network and services should be the same regardless of if it is a fixed or mobile network, or what type of mobile network is connected to. LTE and SAE will allow the mobile network to provide high data rates and low latency to a mobile user at a comparable rate to fixed line users. This will be vital in ensuring the same feel to the services when in the mobile environment.

Network Operators

There is a trend within network operators and service providers to offer "Triple play" and more services, where multiple communications services are provided to a customer from a single subscription. This strategy is to increase Average Revenue Per User (ARPU) and reducing churn by keeping multiple services bundled into a single package. The move to NGN and FMC provides two key advantages to provide this package. Firstly, the integration of the mobility management and data routing means that it is more reliable and convenient to provide this to a customer, as they can achieve this with a single telephone number and single device. Today's triple play solutions still require multiple access devices to provide voice communications across different access technologies. Secondly, there is a lower OPEX (OPerational EXpenditure) as the network is simplified, and this reduced the costs for the operator to provide the services. The operator does not need multiple networks and technologies to be maintained side by side for parallel services. As an example, today's triple play operators typically need to operate and maintain the following types of network:

- IP network for residential/business broadband, using a mix of fibre and copper. With extensions for WLAN and WiMAX hubs.
- Circuit Switched network for residential/business voice, using a mix of fibre and copper.
- ATM network for mobile network infrastructure, using fibres.
- DVB network for broadcast services (TV, radio etc)

This means that they may have separate infra-structure for each that has a set of associated running costs, maintenance costs etc. When combining all of the services to fewer network elements and common network IP technologies, then the network becomes more scalable, and new services are quickly rolled out or coverage increased without major new capital investments required.

By implementing an LTE/SAE network, the operators will be able to simplify the overall network architecture and remove old legacy telecoms equipment that no longer provides competitive and cost efficient use of the radio spectrum resources. At the same time, fixed line services will be made available to mobile users without reduction in performance and feel of the service.

Testing Challenges

RF Testing

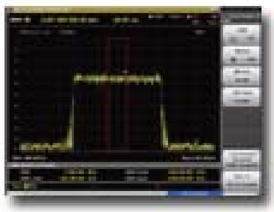
The RF test requirements are given in 3GPP standards TS36.141 Base Station Conformance Test, and TS36.521 UE Conformance Specification Radio transmission and reception. Examples of the common radio transmit characteristics are shown below:

OFDM Radio Testing

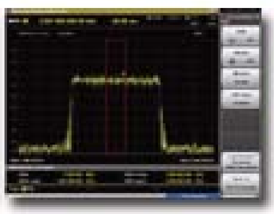
OFDM and the use of high order 64QAM modulation require high linearity, phase and amplitude in both TX and RX modules to prevent inter-symbol interference and to enable accurate IQ demodulation. This requires a fast and adaptive EVM measurement capability to track and measure the signals during adaptive frequency channel use. Testing is made on both the 'per subcarrier' performance of each individual sub-carrier, and then on the 'composite' signal where the sub-carriers are combined and the overall performance is seen.

The sub-carriers require good phase noise performance to prevent leakage across carriers. The frequency mapping and orthogonal properties of OFDM require the 'null' in one carrier phase response being exactly on the peak on the adjacent carrier. Thus, accurate measurement of the phase linearity and amplitude linearity on each sub-carrier is important for the orthogonal design (resistance to interference) of the system.

The OFDM transmissions must also be measured 'per resource block' to see how the power levels in each burst are being correctly maintained. Each individual 'resource element' (that is a single OFDM carrier at one time duration/symbol) has a specific power level to be transmitted, and these power levels should be correctly measured across a whole resource block



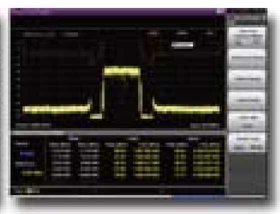
Channel Power



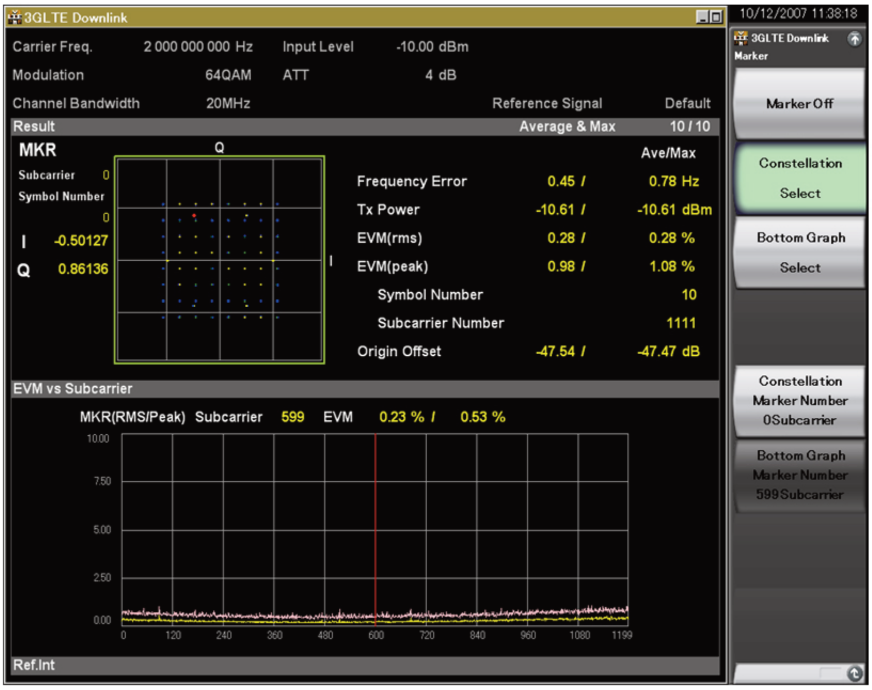
Occupied Bandwidth

ACLR

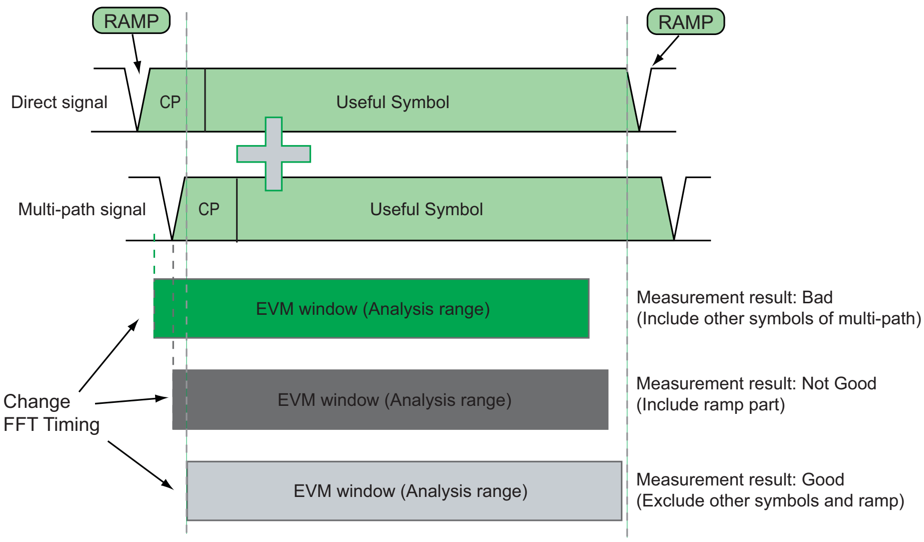
Spurious Emmisions



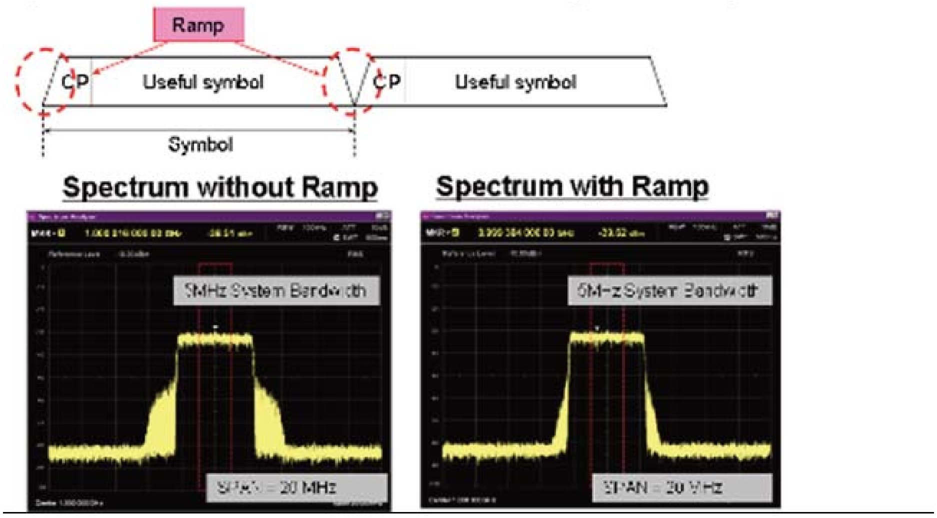
Spectral Emission Mask



The EVM measurement for LTE needs careful consideration because of 2 special features used. Firstly there is the 'Cyclic Prefix' (CP) which is a short burst of transmission at the start of each symbol. This is actually a repeat of the end of the symbol, and is used to give a 'settling time' to allow for delay spread in the transmission path. If measurement is started too soon in the CP period then there will also be signal from the previous symbol (inter-symbol interference, ISI) included that will corrupt the measurement. The second point is that the symbol transmission has a 'ramp' at the start and end to ensure that there is no strong 'burst' of power (a strong burst will create high levels of harmonic distortion and interference). So the start and end of the symbol have an up/down ramp on the power. Therefore we must also restrict the period of the symbol that is measured, to ensure we do not measure in these ramp periods. The technique to address both these issues is to use a 'sliding FFT', where the period of the symbol that is measured can be adjusted in time (sliding) to give the best EVM value. This is shown in the diagram below.



The effect of the 'ramp' can be seen in the measurements below. The waveform on the left has no ramp, and so there is a sharp switch on/off between each symbol. This causes a large 'spectrum due to switching' emission that is seen as broadening of the output spectrum beyond the desired system bandwidth (in this case 5 MHz). The waveform on the right has ramping enabled, and so there is a much less severe switch on/off between symbols. This has the clear effect of reducing spectrum emissions. This type of ramping (also called spectrum shaping) is required to ensure that the transmitter output stays within the allocated frequency band and does not interfere with any adjacent frequencies.



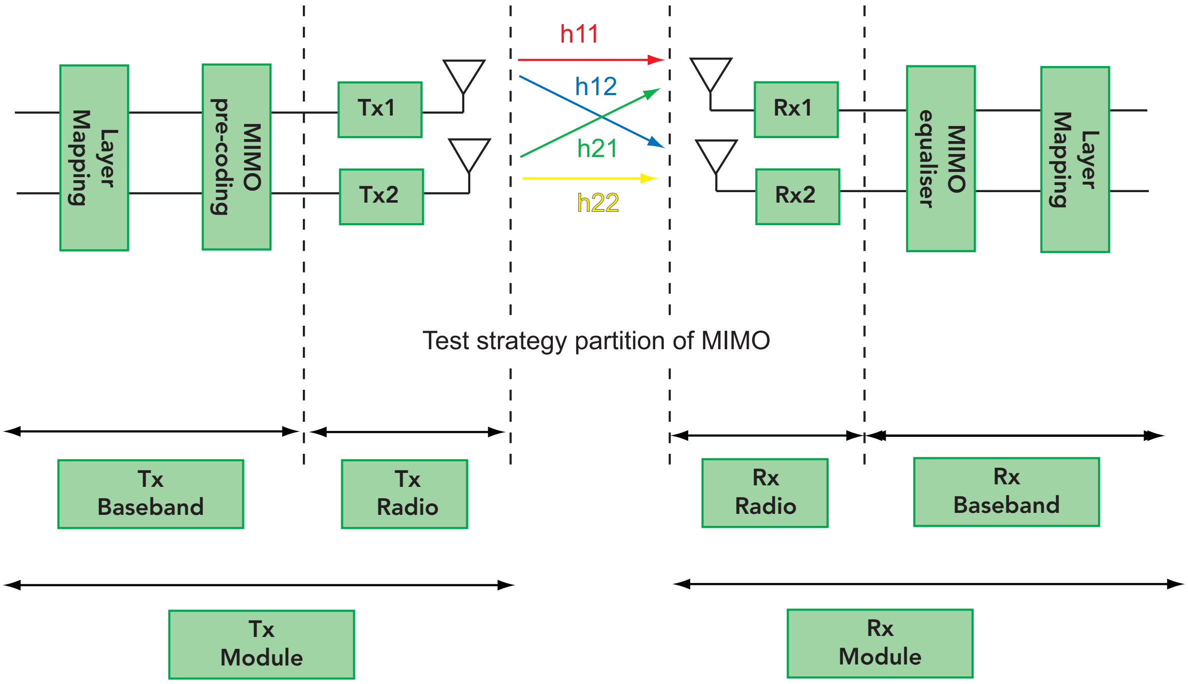
Ramp Processing - The ramp is processed in each symbol to improve spreading of the spectrum waveform due to the discontinuity between symbols.

MIMO Testing

In a MIMO system, the coupling from antenna to air characteristics must be fully understood. Data rate and performance of MIMO links depend on how the multiple RF antennas couple to each other. Accurate calibration of antenna paths, factory calibration and then field installation calibration is required to implement a successful MIMO system. During R&D phase, evaluation of designs is required to confirm the sensitivity calculations required to find critical performance limiting issues.

The base station transmit antenna array may use specialist phased array techniques (like a Butler Matrix) for accurate control of the phase/timing in each antenna path. This requires accurate characterisation of the RF path in terms of electrical path length, coupling and reflections from both ends. This data is then fed into the MIMO adaptation algorithms to enable features such as beam steering. A Vector Network Analyser would normally be used for complete characterisation of the antenna paths.

In testing of MIMO we should consider to test both the baseband processing section and the RF generation/alignment. Both areas need to be tested for both functional test (correct operation) and performance test (optimisation of algorithms for best processing and data throughput efficiency). In addition, it is useful to check the 'negative test', which is to use deliberately incorrect signals and ensure that they are correctly handled or rejected.



A good approach to developing a MIMO test strategy is to lay out a matrix of each of the areas and stages for MIMO test, and then identify the required solutions for each part of the matrix. The key elements of the matrix are to test the individual sub-units (tx baseband, tx radio, rx radio, rx baseband) separately, and then as integrated tx and rx modules. So we can develop a test matrix as below:

	Tx baseband	Tx radio	Tx module	Rx radio	Rx baseband	Rx module
Functional test						
Performance test						
Negative test						

In a MIMO system, it is necessary to calculate the characteristics of the RF path from each TX antenna to each RX antenna. This is required so that the two paths can be separated by the processor and effectively become two separate data paths. To achieve this, the system must accurately measure in real time the RF path characteristics. These algorithms are built into the design of the particular MIMO system used, but they all basically require the accurate phase and amplitude measurement of a 'pre-amble' or 'pilot tone' that is a known signal. For testing environments this provides two challenges:

1. To ensure that the test system can generate accurate reference signals against which the measurements are made. The accuracy of the received signals must be carefully measured, and the test system calibrated to separate the measurement system uncertainty from the MIMO system accuracy and uncertainty. This way, the exact characteristics of the MIMO system are measured, with minimum influence from the test system. This requires the development of a test method/environment to generate the 'reference signals' against which the measurements are made, and then to confirm the measurement method by adjusting the 'quality' of the reference and checking that the measured result matches to the change made.

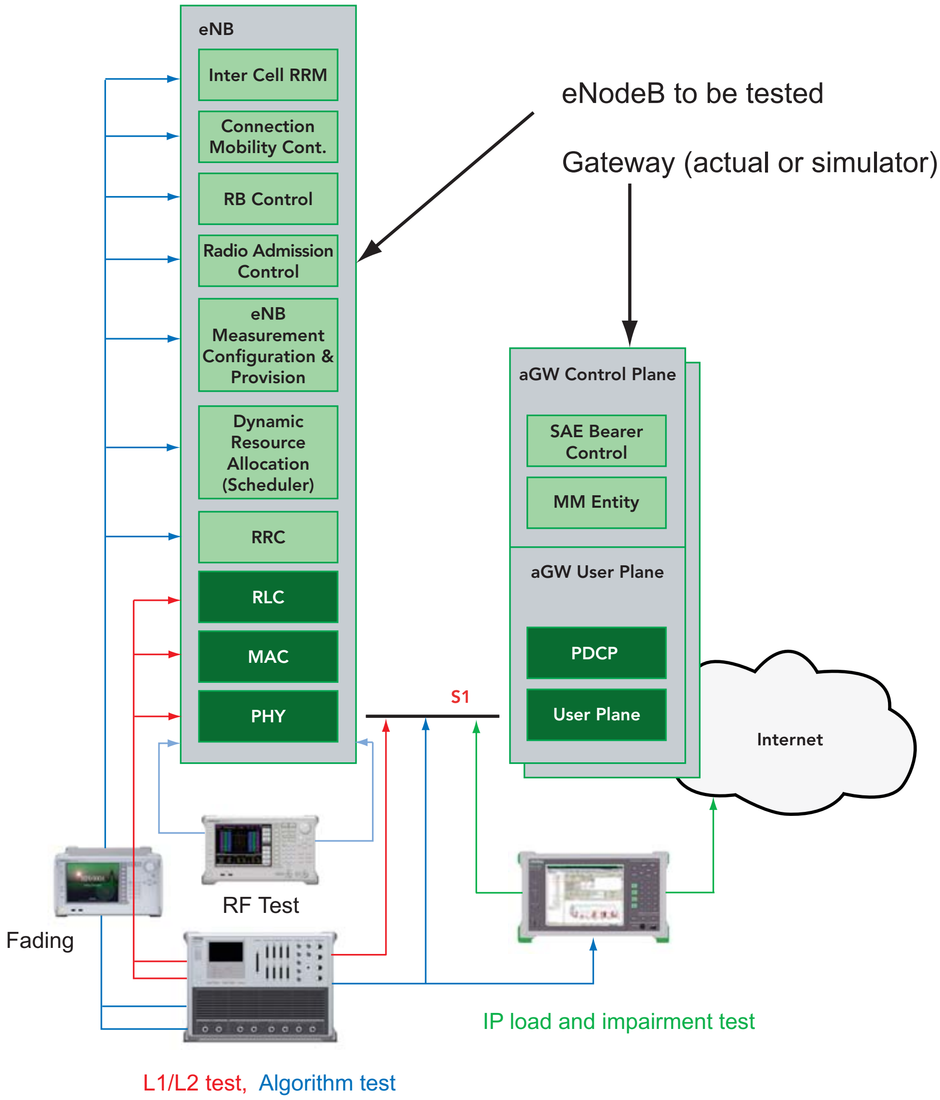
2. The actual RF coupling from transmitter to receiver will affect the measured end to end system performance. So for test environments used in performance measurement, algorithm tuning, Integration and Verification (I&V), and production quality, the RF coupling between antenna must be defined, repeatable and characterised if absolute performance figures (e.g. Mbit/second data rates) are to be measured. This requires the use of suitable fading and multipath test equipment and generation/profiles to create the different coupling between antennas. This can be achieved using static signals (e.g. signal generator based references) for initial checking, and then through baseband fading simulators to verify correct operation at algorithm level, and the finally RF fading simulators for 'end to end' system level testing.

The MIMO coding of data blocks (block coding) is based on Space-Time Block coding, where the actual coding of the data is based on both the space (i.e. which antenna) and time (when it is transmitted), and the diversity gains of MIMO are from using diversity of both space and time for each block of data that is sent. So we must measure both the time alignment of each antenna, and then the spatial alignment of the paths between the antennas.

MIMO test requires extensive test and evaluation of the signal processing and MIMO coding algorithms that are used. To do this, we need a 'step by step' approach where the individual processing and feedback steps in the MIMO algorithms can be isolated and test 'stand alone'. In addition, we need a controlled environment where we can integrate the parts of MIMO algorithm and run them against 'reference conditions' so we can verify the performance. The verification requires that we set up precise know conditions, both with the RF coupling from transmitter to receiver, but also the measurements and feedback reports being made between transmitter and receiver. To do this we require testing at a pure 'baseband' level to check algorithms, as well as testing at RF 'air interface'. In addition we require precise control of the baseband processes and RF coupling. This is normally achieved by using fading simulators and system simulators. The fading simulator will provide a controlled 'air interface' coupling (either real or simulated), and the system simulator will provide a controlled baseband environment (e.g. a controlled UE to test a basestation, or a controlled basestation to test a UE).

When we are including the 'fading' function into the MIMO test, we need to describe fully both the fading of each path (delay paths phase, amplitude and scattering types) and then the correlation between each RF path (the correlation matrix). In the case of '2x2' MIMO there are 4 paths, referred to as h11, h12, h21, and h22, and shown above. In an ideal environment for MIMO there is no correlation between the different RF paths, and so the processing algorithms can fully separate the signals from each path and get the full data rate increase. In the 'real world' we see correlation between the different paths as they have some similar 'shared' routes from transmitter to receiver. In a worst case we have very high correlation between the RF paths as they essentially have the same path. For each of these scenarios we have a 'correlation matrix' that describes mathematically how the different RF paths are related, and then we must test, verify and then optimise the algorithms to give best possible data rate throughputs in each of the different types of RF environment that could be experienced.

An example test environment for an ENodeB is shown below:



LTE eNodeB MIMO Test Environment

L1 Testing

L1 contains the algorithms and procedures associated with reporting and measurements that then drive the Power Control, Adaptive Modulation and Coding, and MIMO processing capabilities. So from a test point of view we need to verify both the correct measurement are made at the receiver (and then transmitted back to the corresponding element that uses the measurement), and then that the transmitter is correctly reacting to the measurement reports and adjusting parameters accordingly.

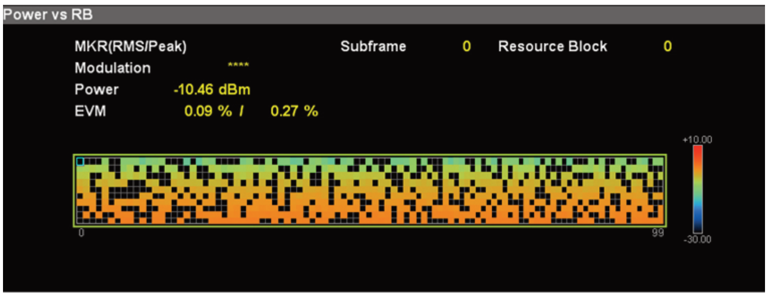
Reports made by the UE back to the network include CQI (Channel Quality Indicator), PMI (Pre-Coding Matrix Indicator), RI (Rank Indicator). CQI is associated with selection of AMC (data rate) in the NodeB, and PMI and RI are used by the NodeB to configure the MIMO encoding.

In addition, the UE must measure the Reference Signal Received Power (RSRP), and so must correctly identify the antenna specific reference signals and measure power in the individual resource elements containing the reference signals.

The NodeB must adjust the Timing Advance of the UE, so that all UE's are received with the same relative timing (required for an efficient FFT process in NodeB receiver). So the NodeB must detect the timing offset of the UE (time difference due to 'time of flight' of the signals that are associated with distance from NodeB, and then timing errors in UE as it tries to lock to NodeB Frame Timing). So it is necessary to test the UE responds correctly to timing advance commands from the Node B. The NodeB must be tested to ensure correct handling of 'out of time' UE signals, and correction processes to adjust the UE's reference timing.

Two typical L1 tests are shown below, firstly the Power versus Resource Block (RB), showing the individual power transmitted in each resource block for a single time period (sub-frame). This will evaluate how the power is distributed across all of the available resource blocks, and hence how the available resources have been set to the correct power level for the receiver based on reports and L1 Power Control algorithms.

Secondly, the Power versus Resource Block (RB) showing the variation in time of each RB. Each RB is measured over each time period (Sub-frame), and power level is shown by the colour of the Resource Block.



L2/L3 Testing in LTE

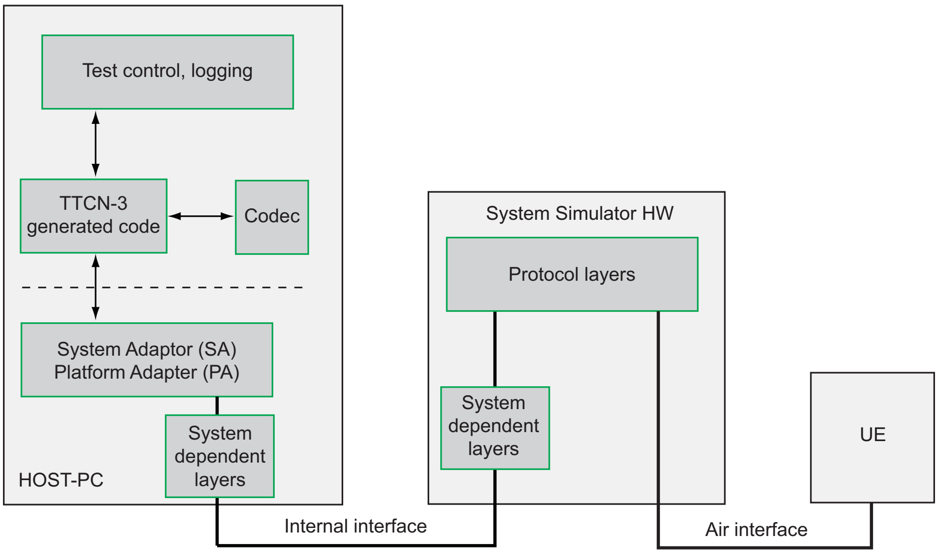
Layer 2 & Layer 3 testing is concerned with the signalling and message flows between the different layers in the protocol stack. In particular, the correct transfer of incoming messages received at higher levels down to/from the physical link layer (Layer 1) circuits and processes, and the correct handling of messages coming back from Layer 1. In addition, L2/L3 has to perform configuration and state control processes to ensure the UE and network are always in the correct state for communication.

This testing is normally made using a 'system simulator' to generate and receive the messages to/from the protocol stack being tested. In addition, the simulator normally has a layer 1 and physical layer implementation so that it can communicate to the target protocol stack through the Layer1 being used. Optionally, the Layer 1 may be omitted and a "virtual layer 1" may be used to link the Layer 2 & Layer 3 elements of the simulator to the protocol stack.

3GPP has defined L1 requirements and test specifications (TS 36.8xx).
3GPP has specified L2/L3 conformance test suites (TS 36.521, TS 36.523).
Common test environments are specified (TS 36.508).
Logical test interface, test loop and test model specified (TS 36.509).

The system simulator is usually one of the following 3 types, dependant on the object being tested:

- Network Simulator, for UE test.
- UE Simulator, for eNodeB test.
- IP Simulator, for Gateway test.



Test System Architecture for UE Test

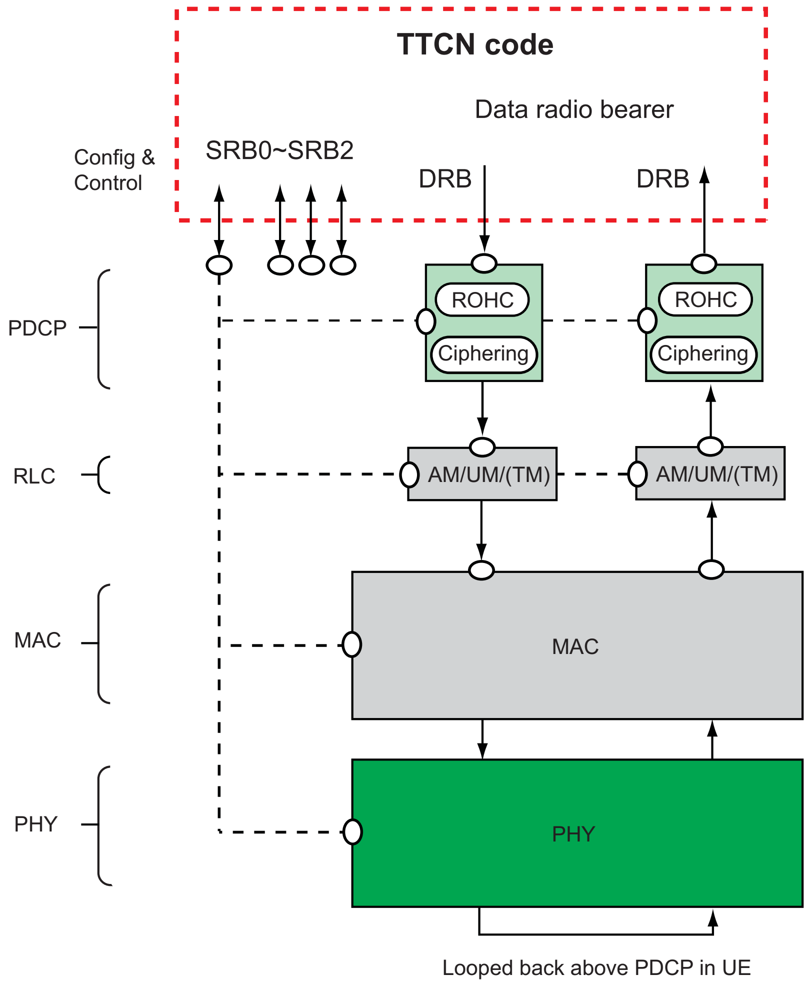
The different simulators all have a similar architecture, using L1 hardware for PHY layer connection, and then a control environment (usually PC hosted) for L2/L3 and logging/analysis. A typical implementation for UE testing is shown below. For LTE UE test, the script for generating the test messages to/from the UE is written in TTCN-3 programming language.

UE Test Loop Modes

It is normally required to configure special test loop modes, where data being received by a device is automatically re-transmitted back by the device. This is to enable the data rates and connectivity to be verified by sending specific data patterns to a device and checking that they are correctly received and re-transmitted. This is not the normal operation mode of the devices, and so this special test loop mode is only activated during device test.

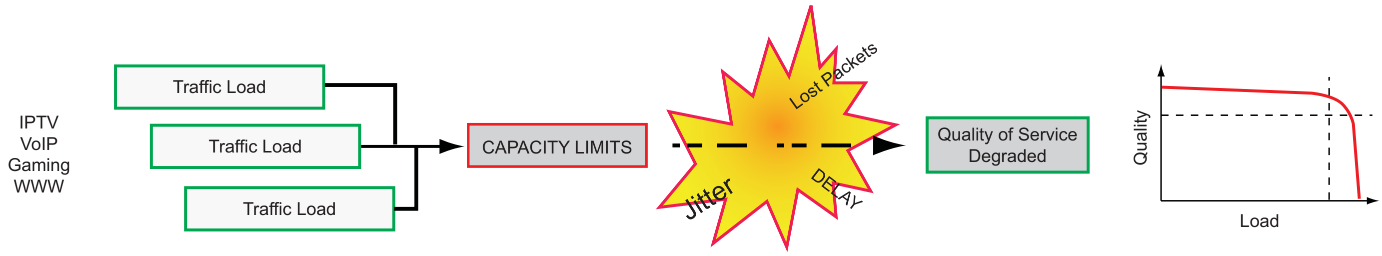
A large number of MAC, RLC, PDCP/ROHC and nearly all Data Radio Bearer (DRB) LTE tests need the test loop in the UE. Without test loop, DRB will not have only limited test coverage and L2 will have in-sufficient test coverage to ensure correct operation. A single test loop at UE is required for signalling test in TTCN3, and the closing loop point in UE is set above PDCP entity. The UE PDCP entities are configured during loop test, ciphering may be on and ROHC possibly configured for DRB test. Ciphering can be set to a dummy algorithm for some tests if necessary.

An example of implementing the functionality into a system simulator is shown below, enabling the simulator to activate the UE into test loop mode and then send test data to/from the UE to verify data throughput and correct operation of the individual signalling layers. Normally, the PHY, MAC, RLC and PDCP functions can all be tested individually and then combined into a throughput and performance test.


TTCN code
Data radio bearer
Config &
Control
SRB0~SRB2
DRB
DRB
PDCP
ROHC
Ciphering
ROHC
Ciphering
RLC
AM/UM/(TM)
AM/UM/(TM)
MAC
MAC
PHY
PHY
Looped back above PDCP in UE

Test Requirements in SAE

SAE is based on an "All IP" network concept, with use of IPv6 to provide the IP protocol. This protocol was developed from traditional 'wired' networks where the common traffic flow problems are due to overload or broken connections. The capacity (bandwidth) of each data link is usually static, and traffic flow problems arise from the link being overloaded or effectively zero (broken cables, damaged routers etc). Capacity issues arises usually from volume of traffic from the users, and the network operator can easily control in a static way how many users are connected to a particular hub and what bandwidth they are offered. In a wireless link, and particularly a fast adaptive link like LTE, the capacity of the data link is variable according to the environment (RF path loss, distance to base-station) or according to loading of the cell (the number of users in the cell can vary in real time, without the control of the operator).



Effects of Load on Traffic - Increasing load reduces quality when capacity threshold is exceeded. Excess delay, excess jitter or lost packets

So the IP routing, flow control and QoS mechanisms must be adaptable to varying bandwidth, where the variations are:

1. Fast (RF fading happens in less than 1 second).
2. Variable (the level of RF loss can vary 20-30dB in a few seconds) causing data rates to vary from 100KB/s to 10MB/s in less than a second.
3. Non predictable (depends on human behaviours of customers in the cell).

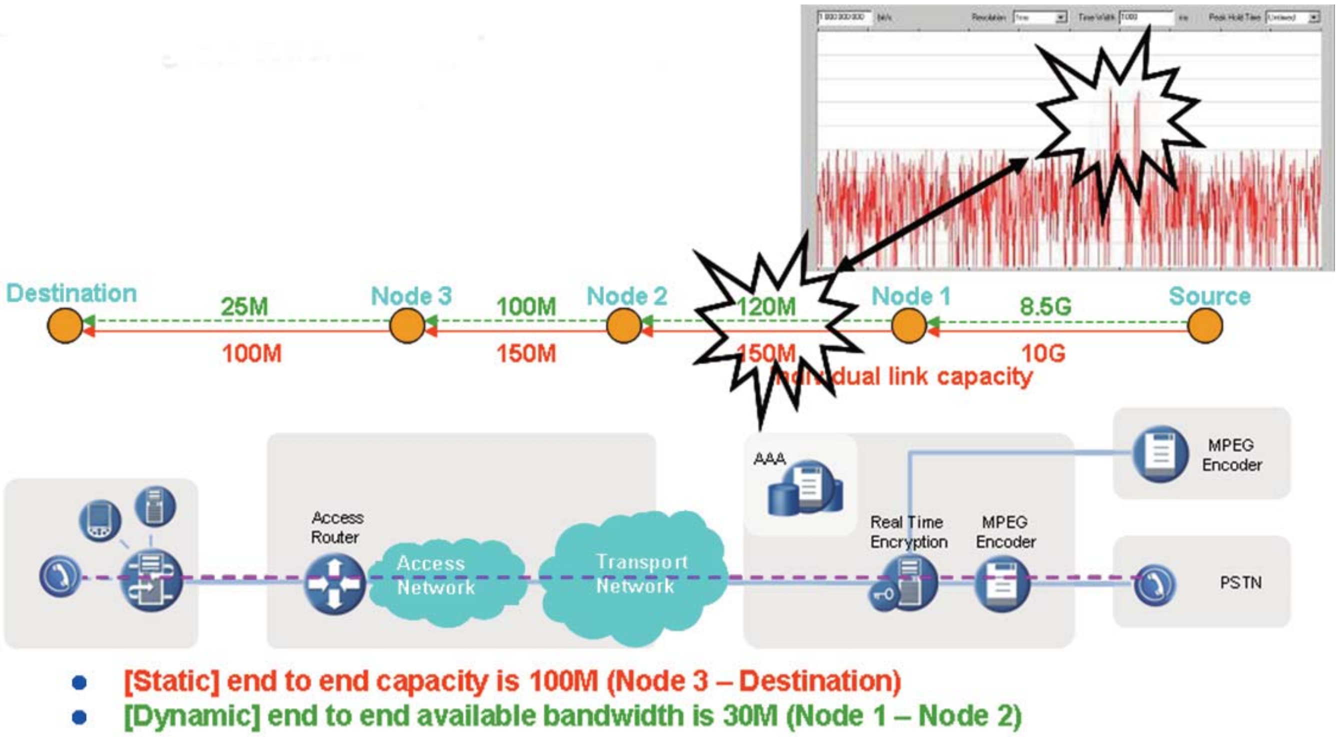
This will require the use of IP technologies suited for variable bit rates and Quality of Service (QoS) provisioning. To manage the fast variable data rates, LTE uses re-transmission technology in the eNB. This requires buffering and flow control mechanisms into the eNB from the core network to prevent data overflow or loss when there are sudden signal fades that require a high amount of re-transmission. So the IP network must be extensively tested for its ability to manage the flow control and re-transmission algorithms.

The types of services offered in the networks can be classified into 4 types of service with the following characteristics:

Class of service	Bandwidth	Latency	QoS Requirement	Example
Conversational	Low-medium	Low	Guaranteed	VoIP/Video calling
Streaming	High	Low	Guaranteed	IPTV, multi-media streaming
Browsing	Low-medium	Normal	Best Effort	Web browser
Background	Medium	Normal	Minimal	Email synchronisation
Broadcast	High	Low	Guaranteed	Multi-cast

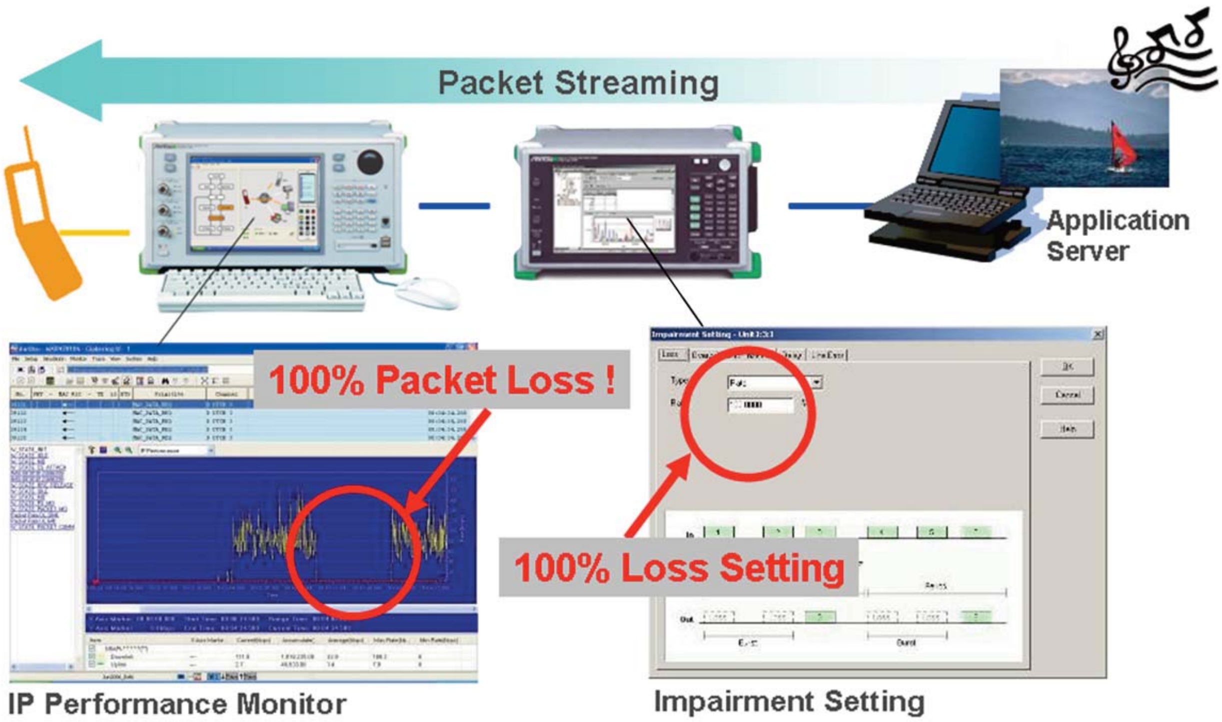
It can be seen from the above that Broadcast services have the most demanding requirements for the network. This is why there is now a strong trend to dedicated Broadcast technologies within a mobile network to meet these requirements (e.g. MBMS, MediaFlo, DVB-H).

The network quality assurance testing and monitoring systems must be able to monitor the delivery of each of the types of service, to ensure the customer expectations are met. This involves firstly using test equipment at the individual node/element level to verify the performance of the IP traffic through the network element, such as load test, latency, jitter etc. Secondly, the performance of the equipment in the live network should be monitored using live traffic flow logging, decode and analysis tools. This will highlight the inter-action between network elements according to parameters such as load, architecture and resource allocations, and allows optimisation of the whole network to meet the customer's expectations.



End to End Available Bandwidth: The Capacity Planning Problem

When we consider the effects of mobility and cellular environments on NGN services, we need to verify that the services can work correctly under the demanding conditions of a wireless network. Typical Internet based applications may have been developed assuming a robust link between client and server, with only limited impairment such as delay and jitter. In an wireless cellular environment such as LTE/SAE , the 'handover', fading and mobility can introduce significant delay and variation in data rates (hence in jitter) that are beyond those normally expected from a wireline network. So, the services developed for a wireline network (e.g. VoIP, Instant Messaging) need to be tested in the more demanding cellular environment. We can make such a test of the applications, client and servers using network simulators and traffic impairment simulators, to provide a controlled and repeatable test environment to isolate these effects and evaluate their impact on user experience.



Mobile Network Impairment Simulation - Loss Insertion Example - Ensure Stability During Wireless Link Loss

3GPP References

36.104 Evolved Universal Terrestrial Radio Access (E-UTRA); Base Station (BS) radio transmission and reception

36.201 Evolved Universal Terrestrial Radio Access (E-UTRA); Long Term Evolution (LTE) physical layer; General description

36.211 Evolved Universal Terrestrial Radio Access (E-UTRA); Physical channels and modulation

36.212 Evolved Universal Terrestrial Radio Access (E-UTRA); Multiplexing and channel coding

36.213 Evolved Universal Terrestrial Radio Access (E-UTRA); Physical layer procedures

36.214 Evolved Universal Terrestrial Radio Access (E-UTRA); Physical layer; Measurements

36.300 Evolved Universal Terrestrial Radio Access (E-UTRA) and Evolved Universal Terrestrial Radio Access (E-UTRAN); Overall description; Stage 2

36.302 Evolved Universal Terrestrial Radio Access (E-UTRA); Services provided by the physical layer

36.304 Evolved Universal Terrestrial Radio Access (E-UTRA); User Equipment (UE) procedures in idle mode

36.321 Evolved Universal Terrestrial Radio Access (E-UTRA); Medium Access Control (MAC) protocol specification

36.322 Evolved Universal Terrestrial Radio Access (E-UTRA); Radio Link Control (RLC) protocol specification

36.323 Evolved Universal Terrestrial Radio Access (E-UTRA); Packet Data Convergence Protocol (PDCP) specification

36.401 Evolved Universal Terrestrial Radio Access Network (E-UTRAN); Architecture description

36.410 Evolved Universal Terrestrial Radio Access Network (E-UTRAN); S1 general aspects and principles

36.411 Evolved Universal Terrestrial Radio Access Network (E-UTRAN); S1 layer 1

36.412 Evolved Universal Terrestrial Radio Access Network (E-UTRAN); S1 signalling transport

36.413 Evolved Universal Terrestrial Access (E-UTRA) ; S1 Application Protocol (S1 AP)

36.414 Evolved Universal Terrestrial Radio Access Network (E-UTRAN); S1 data transport

36.420 Evolved Universal Terrestrial Radio Access Network (E-UTRAN); X2 general aspects and principles

36.421 Evolved Universal Terrestrial Radio Access Network (E-UTRAN); X2 layer 1

36.422 Evolved Universal Terrestrial Radio Access Network (E-UTRAN); X2 signalling transport

36.423 Evolved Universal Terrestrial Radio Access Network (E-UTRAN); X2 Application Protocol (X2AP)

36.424 Evolved Universal Terrestrial Radio Access Network (E-UTRAN); X2 data transport

36.801 Evolved Universal Terrestrial Radio Access (E-UTRA); Measurement Requirements

36.803 Evolved Universal Terrestrial Radio Access (E-UTRA); User Equipment (UE) radio transmission and reception

36.804 Evolved Universal Terrestrial Radio Access (E-UTRA); Base Station (BS) radio transmission and reception

36.938 Improved network controlled mobility between LTE and 3GPP2/mobile WiMAX radio technologies

36.942 Evolved Universal Terrestrial Radio Access (E-UTRA); Radio Frequency (RF) system scenarios

Anritsu

Specifications are subject to change without notice.

Anritsu Corporation
5-1-1 Onna, Atsugi-shi, Kanagawa, 243-8555 Japan
Phone: +81-46-223-1111
Fax: +81-46-296-1264

● U.S.A.
Anritsu Company
1155 East Collins Blvd., Suite 100, Richardson,
TX 75081, U.S.A.
Toll Free: 1-800-267-4878
Phone: +1-972-644-1777
Fax: +1-972-671-1877

● Canada
Anritsu Electronics Ltd.
700 Silver Seven Road, Suite 120, Kanata,
Ontario K2V 1C3, Canada
Phone: +1-613-591-2003
Fax: +1-613-591-1006

● Brazil
Anritsu Eletrônica Ltda.
Praca Amadeu Amaral, 27 - 1 Andar
01327-010-Paraiso-São Paulo-Brazil
Phone: +55-11-3283-2511
Fax: +55-11-3288-6940

● Mexico
Anritsu Company, S.A. de C.V.
Av. Ejército Nacional No. 579 Piso 9, Col. Granada
11520 México, D.F., México
Phone: +52-55-1101-2370
Fax: +52-55-5254-3147

● U.K.
Anritsu EMEA Ltd.
200 Capability Green, Luton, Bedfordshire, LU1 3LU, U.K.
Phone: +44-1582-433200
Fax: +44-1582-731303

● France
Anritsu S.A.
16/18 avenue du Québec-SILIC 720
91961 COURTABOEUF CEDEX, France
Phone: +33-1-60-92-15-50
Fax: +33-1-64-46-10-65

● Germany
Anritsu GmbH
Nemetschek Haus, Konrad-Zuse-Platz 1
81829 München, Germany
Phone: +49-89-442308-0
Fax: +49-89-442308-55

● Italy
Anritsu S.p.A.
Via Elio Vittorini 129, 00144 Roma, Italy
Phone: +39-6-509-9711
Fax: +39-6-502-2425

● Sweden
Anritsu AB
Borgafjordsgatan 13, 164 40 KISTA, Sweden
Phone: +46-8-534-707-00
Fax: +46-8-534-707-30

● Finland
Anritsu AB
Teknobulevardi 3-5, FI-01530 VANTAA, Finland
Phone: +358-20-741-8100
Fax: +358-20-741-8111

● Denmark
Anritsu A/S
Kirkebjerg Allé 90, DK-2605 Brøndby, Denmark
Phone: +45-72112200
Fax: +45-72112210

● Spain
Anritsu EMEA Ltd.
Oficina de Representación en España
Edificio Veganova
Avda de la Vega, n° 1 (edf 8, pl 1, of 8)
28108 ALCOBENDAS - Madrid, Spain
Phone: +34-914905761
Fax: +34-914905762

● Russia
Anritsu EMEA Ltd.
Representation Office in Russia
Tverskaya str. 16/2, bld. 1, 7th floor.
Russia, 125009, Moscow
Phone: +7-495-363-1694
Fax: +7-495-935-8962

● United Arab Emirates
Anritsu EMEA Ltd.
Dubai Liaison Office
P O Box 500413 - Dubai Internet City
Al Thuraya Building, Tower 1, Suit 701, 7th Floor
Dubai, United Arab Emirates
Phone: +971-4-3670352
Fax: +971-4-3688460

● Singapore
Anritsu Pte. Ltd.
60 Alexandra Terrace, #02-08, The Comtech (Lobby A)
Singapore 118502
Phone: +65-6282-2400
Fax: +65-6282-2533

● India
Anritsu Pte. Ltd.
India Branch Office
3rd Floor, Shri Lakshminarayan Niwas, #2726,
HAL 3rd Stage, Bangalore - 560 038, India
Phone: +91-80-4058-1300
Fax: +91-80-4058-1301

● P.R. China (Hong Kong)
Anritsu Company Ltd.
Units 4 & 5, 28th Floor, Greenfield Tower, Concordia Plaza,
No. 1 Science Museum Road, Tsim Sha Tsui East,
Kowloon, Hong Kong
Phone: +852-2301-4980
Fax: +852-2301-3545

● P.R. China (Beijing)
Anritsu Company Ltd.
Beijing Representative Office
Room 2008, Beijing Fortune Building,
No. 5, Dong-San-Huan Bei Road,
Chao-Yang District, Beijing 100004, P.R. China
Phone: +86-10-6590-9230
Fax: +86-10-6590-9235

● Korea
Anritsu Corporation, Ltd.
8F Hyunjuk Building, 832-41, Yeoksam Dong,
Kangnam-ku, Seoul, 135-080, Korea
Phone: +82-2-553-6603
Fax: +82-2-553-6604

● Australia
Anritsu Pty. Ltd.
Unit 21/270 Ferntree Gully Road, Notting Hill,
Victoria 3168, Australia
Phone: +61-3-9558-8177
Fax: +61-3-9558-8255

● Taiwan
Anritsu Company Inc.
7F, No. 316, Sec. 1, Neihu Rd., Taipei 114, Taiwan
Phone: +886-2-8751-1816
Fax: +886-2-8751-1817

Please Contact:

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