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Re:	Call for Contributions: Proposed Contribution to ITU-R WP 8F Regarding IMT-2000 Proposal
Abstract	This contribution is draft for the M.1457 application proposal
Purpose	For discussion and decision
Notice	
Release	

Patent Policy and Procedures	The contributor is familiar with the IEEE 802.16 Patent Policy and Procedures <http://ieee802.org/16/ipr/patents/policy.html>, including the statement "IEEE standards may include the known use of patent(s), including patent applications, provided the IEEE receives assurance from the patent holder or applicant with respect to patents essential for compliance with both mandatory and optional portions of the standard." Early disclosure to the Working Group of patent information that might be relevant to the standard is essential to reduce the possibility for delays in the development process and increase the likelihood that the draft publication will be approved for publication. Please notify the Chair <mailto:chair@wirelessman.org> as early as possible, in written or electronic form, if patented technology (or technology under patent application) might be incorporated into a draft standard being developed within the IEEE 802.16 Working Group. The Chair will disclose this notification via the IEEE 802.16 web site <http://ieee802.org/16/ipr/patents/notices>.

IEEE Std. 802.16e based Mobile WiMAX Candidate RTT Submission

Revision 0.06

Informal Draft for Discussion purposes only

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ATTACHMENT 2
(to Circular-letter 8/LCCE/47)

1 Cover Sheet for Submission of proposed radio transmission technologies for IMT-2000 to ITU

The information listed below will be used for cataloguing radio transmission technologies for IMT-2000 by the ITU and will be posted electronically.

This cover sheet (and additional information, if applicable) should be attached when an evaluation group submits a proposal on radio transmission technologies for IMT-2000.

1. Proponent

a) Name of proponent: ____

b) Proponent category:

ITU-R membership: Yes ___ No ___

Regional/National standards body: Yes ___ (Name:_________________) No ___

Industry group: Yes ___ (Name:_________________) No ___

Other: (Name:_________________)
No ___

c) Contact point

Name:

Organization:

Address:

Tel:

Fax:

Email:

2. Proposal identification

a) Name of the proposed RTTs (list all the names) (if the proponent submits multiple proposals):
____

b) Status of proposal:

Revision ___ (former proposed RTTs name:______________)

New proposal ___

3. Proposed RTT(s) service environment (check as many as appropriate)

Indoor ___ Outdoor to indoor pedestrian ___

Vehicular ___ Satellite ___

4. Attachments

Technology template for each test environment ___

Requirements and objectives template ___

IPR statement ___

Other (any additional inputs which the proponent may consider relevant to the evaluation) ___

5. Has the proposal already been submitted to an evaluation group registered with ITU?

Yes ___ (Name of evaluation group: ______________________, Date of submission:________)

No ___

6. Other information

a) Name of person submitting form: ____________________________

b) Date: ____________

2 Detailed Technology Description

2.1 Introduction

The IEEE 802.16 Working Group defines air interfaces standards for Broadband Wireless Access systems. The IEEE Std 802.16e Amendment [1] along with the base IEEE Std 802.16-2004 [2] provide an air interface for combined fixed and mobile broadband wireless access.
The Mobile WiMAX technology being commercialized in the WiMAX Forum is based on the IEEE Std 802.16e and profiles derived from the mandatory and optional feature set described in the standard. The WiMAX Forum also defines a Mobile WiMAX Radio Access Network (RAN) and End-to-End Systems Architecture for such deployable commercial systems.
The Mobile WiMAX System Profile [3] in figure 1 identifies the mandatory and optional features of the IEEE 802.16 standard that are necessary to build a Mobile WiMAX-compliant air interface that can be certified by the WiMAX Forum. The Mobile WiMAX System Profile enables mobile systems to be configured based on a common base feature set. The Mobile WiMAX system profiles are designed to ensure baseline functionality on terminals and base stations that are fully interoperable. Some elements of the base station profiles are specified as optional to provide additional flexibility for deployment based on specific deployment scenarios that may require different configurations that are either capacity-optimized or coverage-optimized. Initial Mobile WiMAX profiles will cover 5, 7, 8.75, and 10 MHz channel bandwidths for licensed worldwide spectrum allocations in the 2.3 GHz, 2.5 GHz, and 3.5 GHz frequency bands.

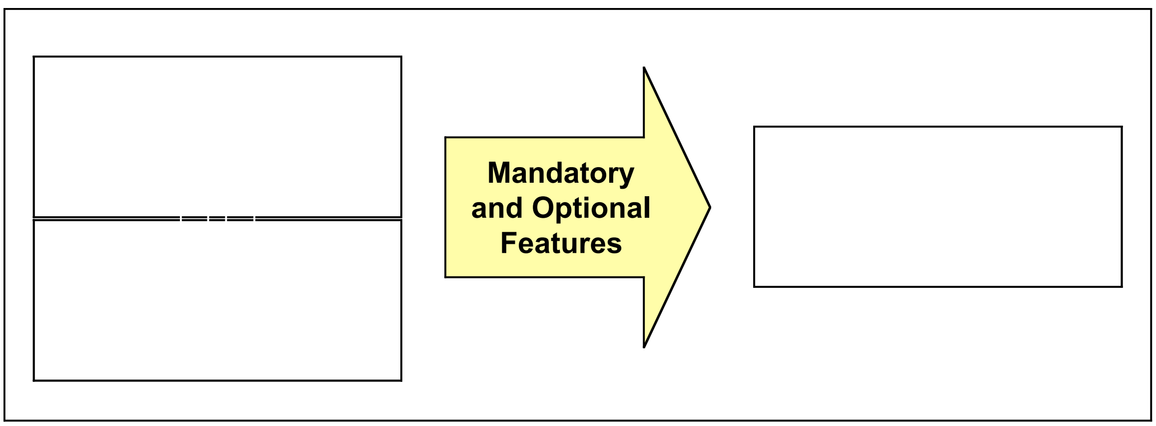

Figure 1: Mobile WiMAX System Profile

Mobile WiMAX systems offer scalability in both radio access technology and network architecture, some of the salient features supported by IEEE Std 802.16e and Mobile WiMAX are:

- **High Data Rates:** The inclusion of MIMO antenna techniques along with flexible sub-channelization schemes, Advanced Coding and Modulation all enable the Mobile WiMAX technology to support peak DL data rates up to 63 Mbps per sector and peak UL data rates up to 28 Mbps per sector in a 10 MHz channel.
- **Quality of Service (QoS):** The fundamental premise of the IEEE 802.16 MAC architecture is QoS. It defines Service Flows which can map to DiffServ code points or MPLS flow labels that enable end-to-end IP based QoS. Additionally, sub-channelization and MAP-based signaling schemes provide a flexible mechanism for optimal scheduling of space, frequency and time resources over the air interface on a frame-by-frame basis.
- **Scalability***:* IEEE Std. 802.16e is designed to be able to scale to work in different channelizations from 1.25 to 20 MHz to comply with varied worldwide requirements as efforts proceed to achieve spectrum harmonization in the longer term. This allows diverse economies to realize the multi-faceted benefits of the

technology for their specific geographic needs such as providing affordable internet access in rural settings versus enhancing the capacity of mobile broadband access in metro and suburban areas.

- **Security:** The features provided for Mobile WiMAX security aspects are best in class with EAP-based authentication, AES-CCM-based authenticated encryption, and CMAC and HMAC based control message protection schemes. Support for a diverse set of user credentials exists including; SIM/USIM cards, Smart Cards, Digital Certificates, and Username/Password schemes based on the relevant EAP methods for the credential type.
- **Mobility:** Mobile WiMAX supports optimized handover schemes with latencies less than 50 milliseconds to ensure real-time applications such as VoIP perform without service degradation. Flexible key management schemes assure that security is maintained during handover.

2.2 Abbreviations

3GPP	3G Partnership Project
3GPP2	3G Partnership Project 2
AAS	Adaptive Antenna System also Advanced Antenna System
ACK	Acknowledge
AES	Advanced Encryption Standard
AG	Absolute Grant
AMC	Adaptive Modulation and Coding
A-MIMO	Adaptive Multiple Input Multiple Output (Antenna)
ASM	Adaptive MIMO Switching
ARQ	Automatic Repeat reQuest
ASN	Access Service Network
ASP	Application Service Provider
BE	Best Effort
CC	Chase Combining (also Convolutional Code)
CCI	Co-Channel Interference
CCM	Counter with Cipher-block chaining Message authentication code
CDF	Cumulative Distribution Function
CINR	Carrier to Interference + Noise Ratio
CMAC	block Cipher-based Message Authentication Code
CP	Cyclic Prefix
CQI	Channel Quality Indicator
CSN	Connectivity Service Network
CSTD	Cyclic Shift Transmit Diversity
CTC	Convolutional Turbo Code
DL	Downlink
EAP	Extensible Authentication Protocol
EESM	Exponential Effective SIR Mapping
EIRP	Effective Isotropic Radiated Power
ErtVR	Extended Real-Time Variable Rate
FBSS	Fast Base Station Switch
FCH	Frame Control Header
FDD	Frequency Division Duplex
FFT	Fast Fourier Transform
FTP	File Transfer Protocol
FUSC	Fully Used Sub-Channel
HARQ	Hybrid Automatic Repeat reQuest

HHO	Hard Hand-Off
HMAC	keyed Hash Message Authentication Code
HO	Hand-Off
HTTP	Hyper Text Transfer Protocol
IE	Information Element
IEFT	Internet Engineering Task Force
IFFT	Inverse Fast Fourier Transform
IR	Incremental Redundancy
ISI	Inter-Symbol Interference
LDPC	Low-Density-Parity-Check
LOS	Line of Sight
MAC	Media Access Control
MAI	Multiple Access Interference
MAN	Metropolitan Area Network
MAP	Media Access Protocol
MBS	Multicast and Broadcast Service
MIMO	Multiple Input Multiple Output (Antenna)
MMS	Multimedia Message Service
MPLS	Multi-Protocol Label Switching
MS	Mobile Station
MSO	Multi-Services Operator
NACK	Not Acknowledge
NAP	Network Access Provider
NLOS	Non Line-of-Sight
NRM	Network Reference Model
nrtPS	Non-Real-Time Packet Service
NSP	Network Service Provider
OFDM	Orthogonal Frequency Division Multiplex
OFDMA	Orthogonal Frequency Division Multiple Access
PER	Packet Error Rate
PF	Proportional Fair (Scheduler)
PKM	Public Key Management
PUSC	Partially Used Sub-Channel
QAM	Quadrature Amplitude Modulation
QPSK	Quadrature Phase Shift Keying
RG	Relative Grant
RR	Round Robin (Scheduler)
RRI	Reverse Rate Indicator
RTG	Receive/transmit Transition Gap
rtPS	Real-Time Packet Service
SDMA	Space (or Spatial) Division (or Diversity) Multiple Access
SF	Spreading Factor
SFN	Single Frequency Network
SGSN	Serving GPRS Support Node
SHO	Soft Hand-Off
SIM	Subscriber Identify Module
SINR	Signal to Interference + Noise Ratio
SISO	Single Input Single Output (Antenna)
SLA	Service Level Agreement

SM	Spatial Multiplexing
SMS	Short Message Service
SNIR	Signal to Noise + Interference Ratio
SNR	Signal to Noise Ratio
S-OFDMA	Scalable Orthogonal Frequency Division Multiple Access
SS	Subscriber Station
STC	Space Time Coding
TDD	Time Division Duplex
TEK	Traffic Encryption Key
TTG	Transmit/receive Transition Gap
TTI	Transmission Time Interval
TU	Typical Urban (as in channel model)
UE	User Equipment
UGS	Unsolicited Grant Service
UL	Uplink
UMTS	Universal Mobile Telephone System
USIM	Universal Subscriber Identify Module
VoIP	Voice over Internet Protocol
VPN	Virtual Private Network
VSF	Variable Spreading Factor
WiFi	Wireless Fidelity
WiMAX	Worldwide Interoperability for Microwave Access

2.3 System Network Architecture

The IEEE air interface Std. 802.16e based Mobile WiMAX End-to-End System Architecture is based an All-IP platform, with support for only IP based packet services and no support for legacy circuit switched services. It offers the advantage of reduced total cost of ownership during the lifecycle of a WiMAX network deployment. The use of All-IP means that a common network core can be used, without the need to maintain both packet and circuit core networks. This results in lower cost, high scalability, and rapid deployment since the networking functionality is primarily expected to be based on software services.

In order to deploy successful and operational commercial systems, there is need for support beyond 802.16 (PHY/MAC) air interface specifications. The Mobile WiMAX End-to-End WiMAX system architecture [5] describes the upper layer specifications for the Radio Access Network and Core Network.

2.3.1 Architecture Principles

The following basic tenets have guided the WiMAX Network architecture development.

1. The architecture is based on a packet-switched framework, including native procedures based on the IEEE 802.16 standard and its amendments, appropriate IETF RFCs and Ethernet standards.
2. The architecture permits decoupling of access architecture (and supported topologies) from connectivity IP service. Network elements of the connectivity system are agnostic to the IEEE 802.16 radio specifics.
3. The architecture allows modularity and flexibility to accommodate a broad range of deployment options such as:
   - Small-scale to large-scale (sparse to dense radio coverage and capacity) WiMAX networks
   - Urban, suburban, and rural radio propagation environments

- Licensed and/or licensed-exempt frequency bands
- Hierarchical, flat, or mesh topologies, and their variants
- Co-existence of fixed, nomadic, portable and mobile usage models

Support for Services and Applications: The end-to-end architecture includes the support for: a) Voice, multimedia services and other mandated regulatory services such as emergency services and lawful interception, b) Access to a variety of independent Application Service Provider (ASP) networks in an agnostic manner, c) Mobile telephony communications using VoIP, d) Support interfacing with various interworking and media gateways permitting delivery of incumbent/legacy services translated over IP (for example, SMS over IP, MMS, WAP) to WiMAX access networks and e) Support delivery of IP Broadcast and Multicast services over WiMAX access networks.

Interworking and Roaming is another key strength of the End-to-End Network Architecture with support for a number of deployment scenarios. In particular, there will be support of a) Loosely-coupled interworking with existing wireless networks such as 3GPP and 3GPP2 or existing wireline networks such as DSL and MSO, with the interworking interface(s) based on a standard IETF suite of protocols, b) Global roaming across WiMAX operator networks, including support for credential reuse, consistent use of AAA for accounting and billing, and consolidated/common billing and settlement, c) A variety of user authentication credential formats such as username/password, digital certificates, subscriber identify modules (SIM/USIM, R-UIM).

2.4 WiMAX Network Reference Model

WiMAX Forum identifies a WiMAX Network Reference Model (NRM) that is a logical representation of the network architecture. The NRM identifies functional entities and reference points over which interoperability is achieved between functional entities. The architecture has been developed with the objective of providing unified support of functionality needed in a range of network deployment models and usage scenarios (ranging from fixed – nomadic – portable – simple mobility – to fully mobile subscribers).

Figure 2 illustrates the NRM, consisting of the following logical entities: MS, ASN, and CSN and clearly identified reference points for interconnection of the logical entities. The figure depicts the key normative reference points R1-R5. Each of the entities, MS, ASN and CSN represent a grouping of functional entities. Each of these functions may be realized in a single physical device or may be distributed over multiple physical devices. The grouping and distribution of functions into physical devices within a functional entity (such as ASN) is an implementation choice; a manufacturer may choose any physical implementation of functions, either individually or in combination, as long as the implementation meets the functional and interoperability requirements.

The intent of the NRM is to allow multiple implementation options for a given functional entity, and yet achieve interoperability among different realizations of functional entities. Interoperability is based on the definition of communication protocols and data plane treatment between functional entities to achieve an overall end-to-end function, for example, security or mobility management. Thus, the functional entities on either side of a reference point represent a collection of control and bearer plane end-points.

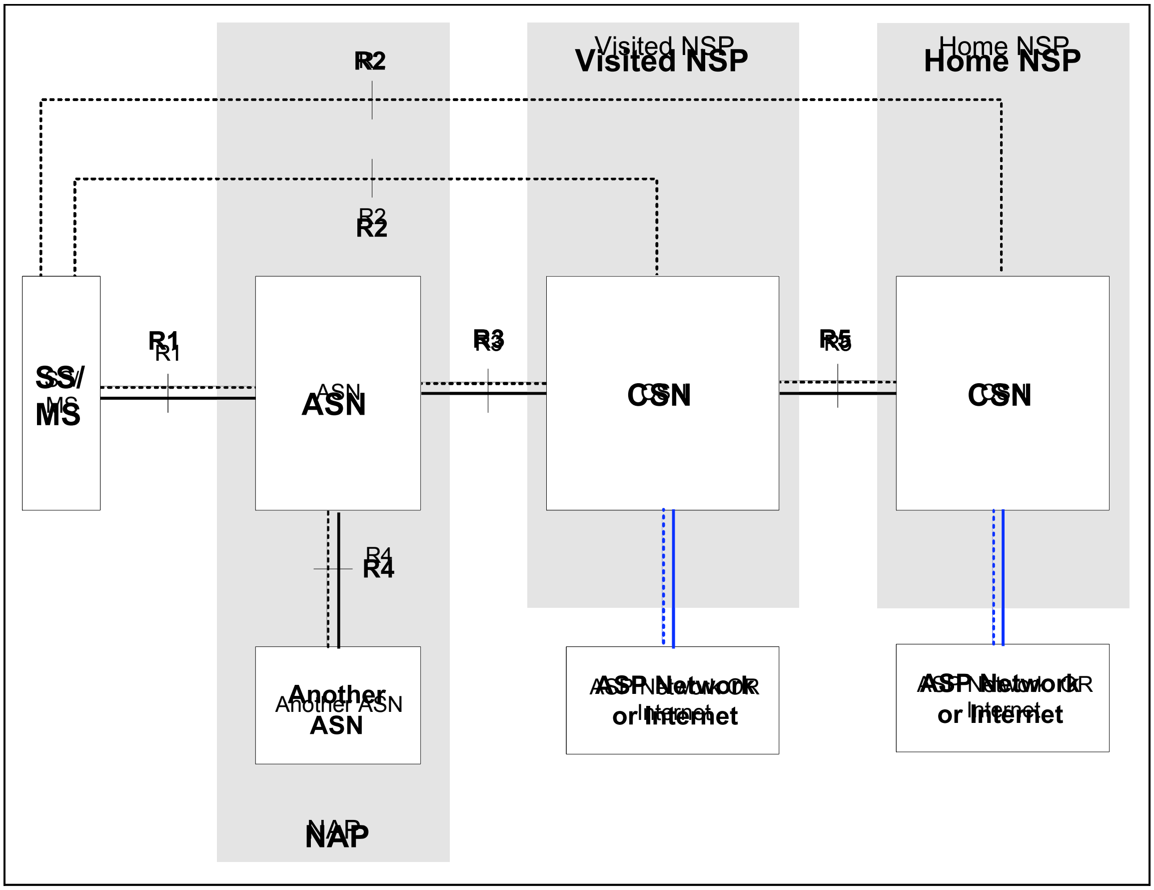

Figure 2: WiMAX Network Reference Model

The ASN defines a logical boundary and represents a convenient way to describe aggregation of functional entities and corresponding message flows associated with the access services. The ASN represents a boundary for functional interoperability with WiMAX clients, WiMAX connectivity service functions and aggregation of functions embodied by different vendors. Mapping of functional entities to logical entities within ASNs as depicted in the NRM may be performed in different ways. The WiMAX Forum is in the process of network specifications in a manner that would allow a variety of vendor implementations that are interoperable and suited for a wide diversity of deployment requirements.

Connectivity Service Network (CSN) is defined as a set of network functions that provide IP connectivity services to the WiMAX subscriber(s). A CSN may comprise network elements such as routers, AAA proxy/servers, user databases and Interworking gateway devices. A CSN may be deployed as part of a Greenfield WiMAX Network Service Provider (NSP) or as part of an incumbent WiMAX NSP.

Figure 3 provides a more basic view of the many entities within the functional groupings of ASN and CSN.

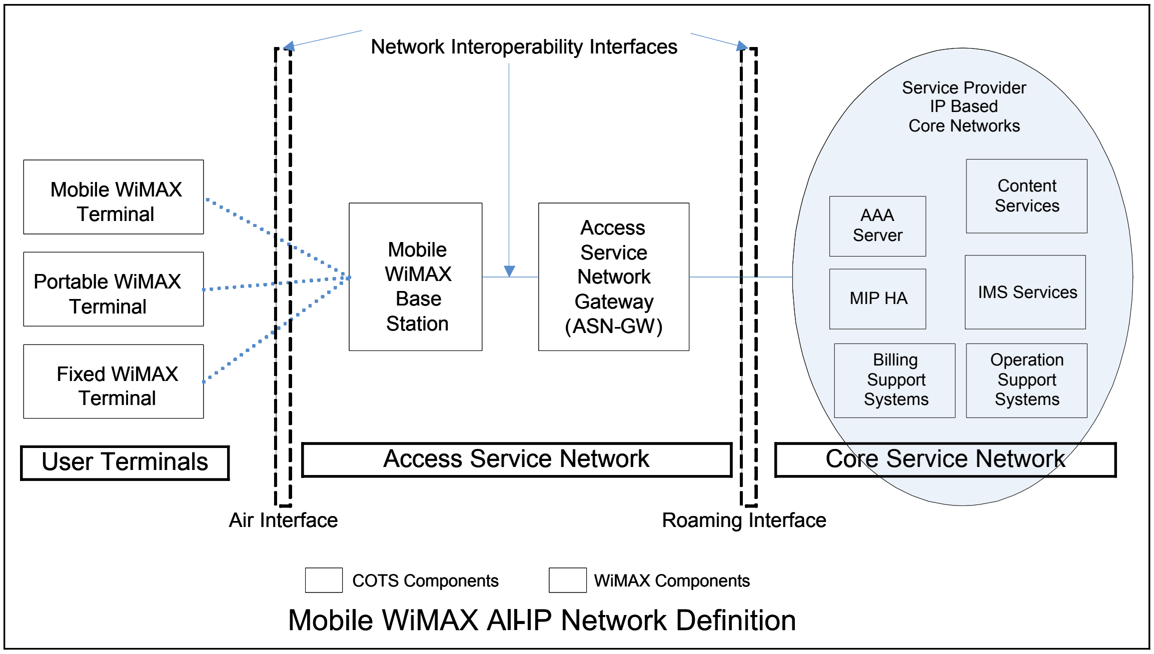

Figure 3: WiMAX Network IP-Based Architecture

The network specifications for WiMAX-based systems are based on several basic network architecture tenets, including those listed below.

Some general tenets have guided the development of Mobile WiMAX Network Architecture and include the following: a) Provision of logical separation between such procedures and IP addressing, routing and connectivity management procedures and protocols to enable use of the access architecture primitives in standalone and inter-working deployment scenarios, b) Support for sharing of ASN(s) of a Network Access Provider (NAP) among multiple NSPs, c) Support of a single NSP providing service over multiple ASN(s) – managed by one or more NAPs, d) Support for the discovery and selection of accessible NSPs by an MS or SS, e) Support of NAPs that employ one or more ASN topologies, f) Support of access to incumbent operator services through internetworking functions as needed, g) Specification of open and well-defined reference points between various groups of network functional entities (within an ASN, between ASNs, between an ASN and a CSN, and between CSNs), and in particular between an MS, ASN and CSN to enable multi-vendor interoperability, h) Support for evolution paths between the various usage models subject to reasonable technical assumptions and constraints, i) Enabling different vendor implementations based on different combinations of functional entities on physical network entities, as long as these implementations comply with the normative protocols and procedures across applicable reference points, as defined in the network specifications and j) Support for the most trivial scenario of a single operator deploying an ASN together with a limited set of CSN functions, so that the operator can offer basic Internet access service without consideration for roaming or interworking.

The WIMAX architecture also allows both IP and Ethernet services, in a standard mobile IP compliant network. The flexibility and interoperability supported by the WiMAX network provides operators with a multi-vendor low cost implementation of a WiMAX network even with a mixed deployment of distributed and centralized ASN's in the network. The WiMAX network has the following major features:

Security

The end-to-end WiMAX Network Architecture is based on a security framework that is agnostic to the operator type and ASN topology and applies consistently across Greenfield and internetworking deployment models and usage scenarios. In particular there is support for: a) Strong mutual device authentication between an

MS and the WiMAX network, based on the IEEE 802.16 security framework, b) All commonly deployed authentication mechanisms and authentication in home and visited operator network scenarios based on a consistent and extensible authentication framework, c) Data integrity, replay protection, confidentiality and non-repudiation using applicable key lengths, d) Use of MS initiated/terminated security mechanisms such as Virtual Private Networks (VPNs), e) Standard secure IP address management mechanisms between the MS/SS and its home or visited NSP.

Mobility and Handovers

The end-to-end WiMAX Network Architecture has extensive capability to support mobility and handovers. It will: a) Include vertical or inter-technology handovers— e.g., to Wi-Fi, 3GPP, 3GPP2, DSL, or MSO – when such capability is enabled in multi-mode MS, b) Support IPv4 or IPv6 based mobility management. Within this framework, and as applicable, the architecture SHALL accommodate MS with multiple IP addresses and simultaneous IPv4 and IPv6 connections, c) Support roaming between NSPs, d) Utilize mechanisms to support seamless handovers at up to vehicular speeds— satisfying well-defined (within WiMAX Forum) bounds of service disruption. Some of the additional capabilities in support of mobility include the support of: i) Dynamic and static home address configurations, ii) Dynamic assignment of the Home Agent in the service provider network as a form of route optimization, as well as in the home IP network as a form of load balancing and iii) Dynamic assignment of the Home Agent based on policies.

Scalability, Extensibility, Coverage and Operator Selection

The end-to-end WiMAX Network Architecture has extensive support for scalable, extensible operation and flexibility in operator selection. In particular, it will: a) enable a user to manually or automatically select from available NAPs and NSPs, b) Enable ASN and CSN system designs that easily scale upward and downward – in terms of coverage, range or capacity, c) Accommodate a variety of ASN topologies - including hub-and-spoke, hierarchical, and/or multi-hop interconnects, d) Accommodate a variety of backhaul links, both wireline and wireless with different latency and throughput characteristics, e) Support incremental infrastructure deployment, f) Support phased introduction of IP services that in turn scale with increasing number of active users and concurrent IP services per user, g) Support the integration of base stations of varying coverage and capacity - for example, pico, micro, and macro base stations and e) Support flexible decomposition and integration of ASN functions in ASN network deployments in order to enable use of load balancing schemes for efficient use of radio spectrum and network resources.

Additional features pertaining to manageability and performance of WiMAX Network Architecture include: a) Support a variety of online and offline client provisioning, enrollment, and management schemes based on open, broadly deployable, IP-based, industry standards, b) Accommodation of Over-The-Air (OTA) services for MS terminal provisioning and software upgrades, and c) Accommodation of use of header compression/suppression and/or payload compression for efficient use of the WiMAX radio resources.

Multi-Vendor Interoperability

Another key aspect of the WiMAX Network Architecture is the support of interoperability between equipment from different manufacturers within an ASN and across ASNs. Such interoperability will include interoperability between: a) BS and backhaul equipment within an ASN, and b) Various ASN elements (possibly from different vendors) and CSN, with minimal or no degradation in functionality or capability of the ASN.

The IEEE 802.16 standard defines multiple convergence sub-layers. The WiMAX Network Architecture framework supports a variety of CS types including: Ethernet CS, IPv4 CS and IPv6 CS.

Quality of Service

The WiMAX Network Architecuture has provisions for support of QoS mechanisms. In particular, it enables flexible support of simultaneous use of a diverse set of IP services. The architecture supports: a) Differentiated levels of QoS - coarse-grained (per user/terminal) and/or fine-grained (per service flow per user/terminal), b) Admission control, c) Bandwidth management and d) Implementation of policies as defined by various operators for QoS-based on their SLAs (including policy enforcement per user and user group as well as factors such as location, time of day, etc.). Extensive use is made of standard IETF mechanisms for managing policy definition and policy enforcement between operators.

2.5 Physical Layer Description

2.5.1 OFDMA Basics

Orthogonal Frequency Division Multiplexing (OFDM) is a multiplexing technique that subdivides the bandwidth into multiple frequency sub-carriers as shown in Figure 4. In an OFDM system, the input data stream is divided into several parallel sub-streams of reduced data rate (thus increased symbol duration) and each sub-stream is modulated and transmitted on a separate orthogonal sub-carrier. The increased symbol duration improves the robustness of OFDM to delay spread. Furthermore, the introduction of the cyclic prefix (CP) can completely eliminate Inter-Symbol Interference (ISI) as long as the CP duration is longer than the channel delay spread. The CP is typically a repetition of the last samples of data portion of the block that is appended to the beginning of the data payload as shown in Figure 5. The CP prevents inter-block interference and makes the channel appear circular and permits low-complexity frequency domain equalization. A perceived drawback of CP is that it introduces overhead, which effectively reduces bandwidth efficiency. While the CP does reduce bandwidth efficiency somewhat, the impact of the CP is similar to the "roll-off factor" in raised-cosine filtered single-carrier systems. Since OFDM has a very sharp, almost "brick-wall" spectrum, a large fraction of the allocated channel bandwidth can be utilized for data transmission, which helps to moderate the loss in efficiency due to the cyclic prefix.

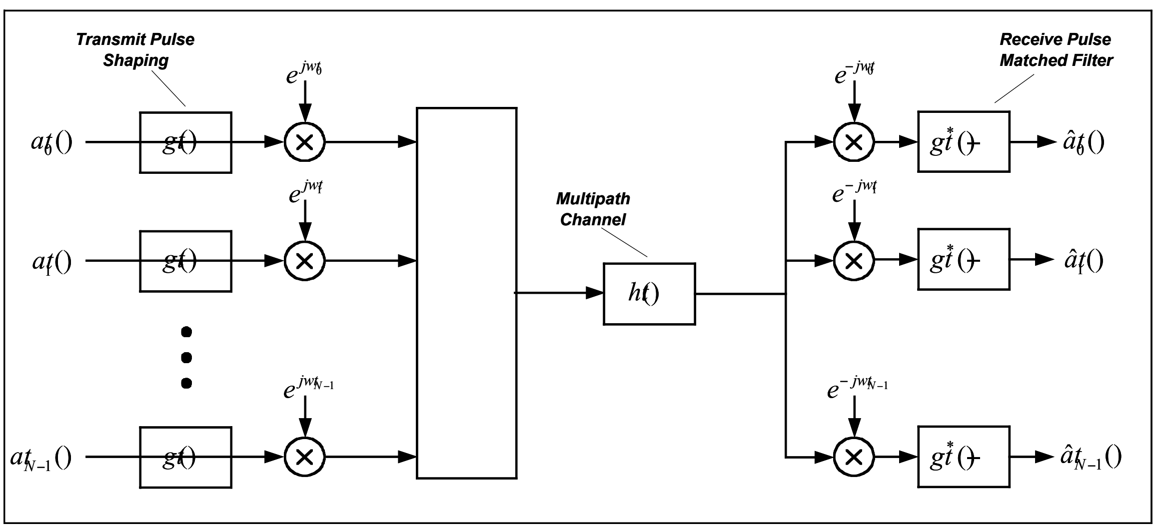

Figure 4: Basic Architecture of an OFDM System

OFDM exploits the frequency diversity of the multipath channel by coding and interleaving the information across the sub-carriers prior to transmissions. OFDM modulation can be realized with efficient Inverse Fast Fourier Transform (IFFT), which enables a large number of sub-carriers (up to 2048) with low complexity. In an OFDM system, resources are available in the time domain by means of OFDM symbols and in the frequency domain by means of sub-carriers. The time and frequency resources can be organized into sub-channels for allocation to individual users. Orthogonal Frequency Division Multiple Access (OFDMA) is a multiple-access/multiplexing scheme that provides multiplexing operation of data streams from multiple users onto the downlink sub-channels and uplink multiple access by means of uplink sub-channels.

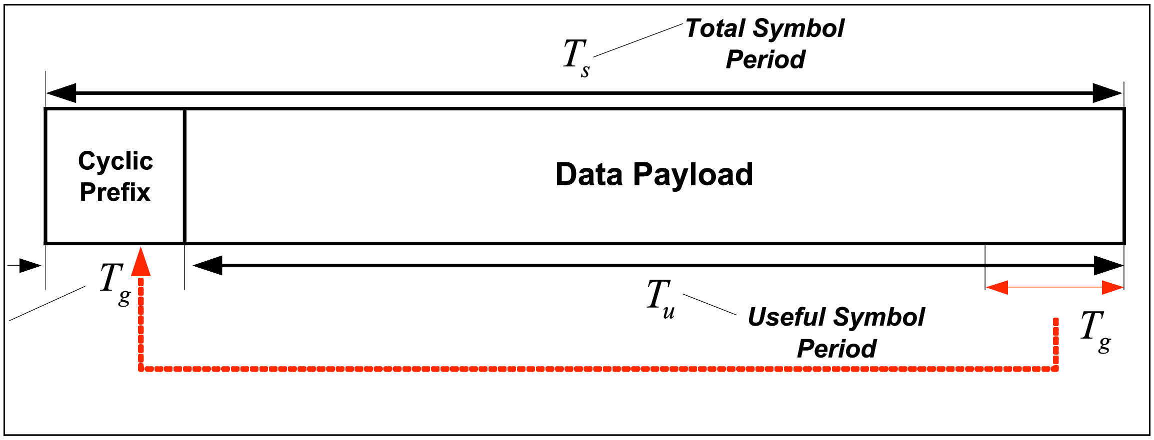

Figure 5: Insertion of Cyclic Prefix (CP)

2.5.2 OFDMA Symbol Structure and Sub-Channelization

The OFDMA symbol structure consists of three types of sub-carriers as shown in Figure 6:

- Data sub-carriers for data transmission
- Pilot sub-carriers for estimation and synchronization purposes
- Null sub-carriers for no transmission; used for guard bands and DC carriers

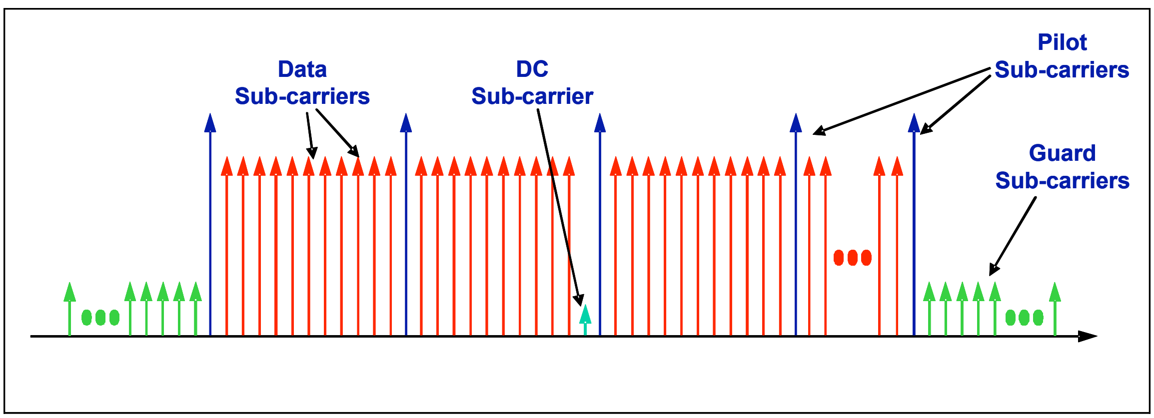

Figure 6: OFDMA Sub-Carrier Structure

Active (data and pilot) sub-carriers are grouped into subsets of sub-carriers called sub-channels. The WiMAX OFDMA PHY [1] supports sub-channelization in both DL and UL. The minimum frequency-time resource unit of sub-channelization is one slot, which is equal to 48 data tones (sub-carriers).

There are two types of sub-carrier permutations for sub-channelization; *diversity* and *contiguous*. The diversity permutation draws sub-carriers pseudo-randomly to form a sub-channel. It provides frequency diversity and inter-cell interference averaging. The diversity permutations include DL FUSC (Fully Used Sub-Carrier), DL PUSC (Partially Used Sub-Carrier) and UL PUSC and additional optional permutations. With DL PUSC, for each pair of OFDM symbols, the available or usable sub-carriers are grouped into *clusters* containing 14 contiguous sub-carriers per symbol, with pilot and data allocations in each cluster in the even and odd symbols as shown in Figure 7.

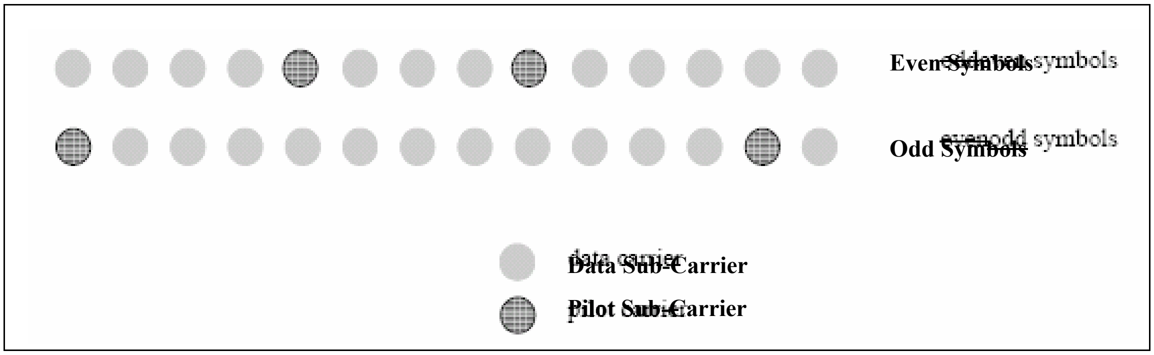

Figure 7: DL Frequency Diverse Sub-Channel

A re-arranging scheme is used to form *groups* of clusters such that each group is made up of clusters that are distributed throughout the sub-carrier space. A sub-channel in a group contains two (2) clusters and is comprised of 48 data sub-carriers and eight (8) pilot sub-carriers. The data subcarriers in each group is further permuted to generate subchannels in the group. Therefore, only the pilot positions in the cluster as shown in Figure 5. The data subcarriers in the cluster are distributed to multiple subchannels.

Analogous to the cluster structure for DL, a *tile* structure is defined for the UL PUSC whose format is shown in Figure 8.

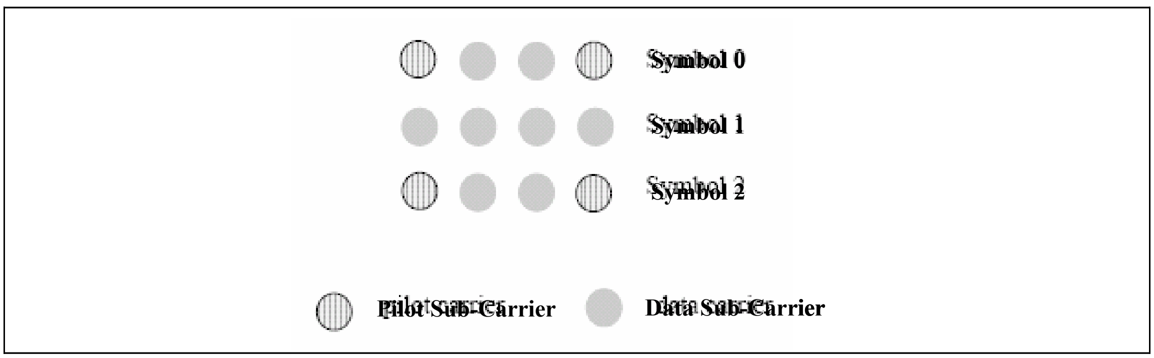

Figure 8: Tile Structure for UL PUSC

The available sub-carrier space is split into tiles and six (6) tiles, chosen from across the entire spectrum by means of a re-arranging/permutation scheme, are grouped together to form a *slot. The slot is comprised of 48 data sub-carriers and 24 pilot sub-carriers in 3 OFDM symbols.*

The contiguous permutation groups a block of contiguous sub-carriers to form a sub-channel. The contiguous permutations include DL AMC and UL AMC, and have the same structure. A bin consists of 9 contiguous sub-carriers in a symbol, with 8 assigned for data and one assigned for a pilot. A *slot* in AMC is defined as a collection of bins of the type (N x $M = 6$), where N is the number of contiguous bins and M is the number of contiguous symbols. Thus the allowed combinations are [(6 bins, 1 symbol), (3 bins, 2 symbols), (2 bins, 3 symbols), (1 bin, 6 symbols)]. AMC permutation enables multi-user diversity by choosing the sub-channel with the best frequency response.

In general, diversity sub-carrier permutations perform well in mobile applications while contiguous sub-carrier permutations are well suited for fixed, portable, or low mobility environments. These options enable the system designer to trade-off mobility for throughput.

2.5.3 Scalable OFDMA

The IEEE 802.16e Wireless MAN OFDMA mode is based on the concept of scalable OFDMA (S-OFDMA). S-OFDMA supports a wide range of bandwidths to flexibly address the need for various spectrum allocation and usage model requirements. The scalability is supported by adjusting the FFT size while fixing the sub-carrier frequency spacing at 10.94 kHz. Since the resource unit sub-carrier bandwidth and symbol duration is fixed, the impact to higher layers is minimal when scaling the bandwidth. The S-OFDMA parameters are listed in Table 1. The system bandwidths for the initial planned profiles being developed by the WiMAX Forum Technical Working Group for Release-1 are 5 and 10 MHz.

Parameters	Values	
System Channel Bandwidth (MHz)	5	10
Sampling Frequency (F_p in MHz)	5.6	11.2
FFT Size (N_{FFT})	512	1024
Number of Sub-Channels	8	16
Sub-Carrier Frequency Spacing	10.94 kHz	
Useful Symbol Time (T_b = 1/f)	91.4 microseconds	
Guard Time ($T_g = T_b/8$)	11.4 microseconds	
OFDMA Symbol Duration ($T_s = T_b + T_g$)	102.9 microseconds	
Number of OFDMA Symbols (5 ms Frame)	48	

Table 1: OFDMA Scalability Parameters

2.5.4 TDD Frame Structure

The 802.16e PHY [1] supports TDD, FDD, and Half-Duplex FDD operation; however the initial release of Mobile WiMAX certification profiles will only include TDD. With ongoing releases, FDD profiles will be considered by the WiMAX Forum to address specific market opportunities where local spectrum regulatory requirements either prohibit TDD or are more suitable for FDD deployments. To counter interference issues, TDD does require system-wide synchronization; nevertheless, TDD is the preferred duplexing mode for the following reasons:

- TDD enables adjustment of the downlink/uplink ratio to efficiently support asymmetric downlink/uplink traffic, while with FDD, downlink and uplink always have fixed and generally, equal DL and UL bandwidths.
- TDD assures channel reciprocity for better support of link adaptation, MIMO and other closed loop advanced antenna technologies.
- Unlike FDD, which requires a pair of channels, TDD only requires a single channel for both downlink and uplink providing greater flexibility for adaptation to varied global spectrum allocations.
- Transceiver designs for TDD implementations are less complex and therefore less expensive.

Figure 9 illustrates the OFDM frame structure for a Time Division Duplex (TDD) implementation. Each frame is divided into DL and UL sub-frames separated by Transmit/Receive and Receive/Transmit Transition Gaps (TTG and RTG, respectively) to prevent DL and UL transmission collisions. In a frame, the following control information is used to ensure optimal system operation:

- **Preamble:** The preamble, used for synchronization, is the first OFDM symbol of the frame.
- **Frame Control Head (FCH):** The FCH follows the preamble. It provides the frame configuration information such as MAP message length and coding scheme and usable sub-channels.
- **DL-MAP and UL-MAP:** The DL-MAP and UL-MAP provide sub-channel allocation and other control information for the DL and UL sub-frames respectively.
- **UL Ranging:** The UL ranging sub-channel is allocated for mobile stations (MS) to perform closed-loop time, frequency, and power adjustment as well as bandwidth requests.
- **UL CQICH:** The UL CQICH channel is allocated for the MS to feedback channel-state information.
- **UL ACK:** The UL ACK is allocated for the MS to feedback DL HARQ acknowledgement.

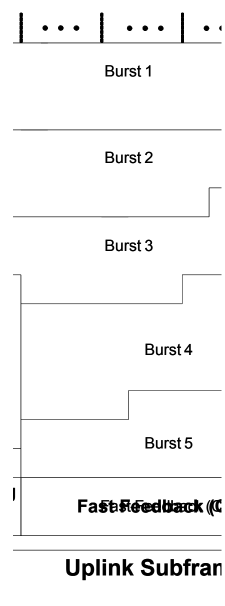

Figure 9: WiMAX OFDMA Frame Structure

2.5.5 Other Advanced PHY Layer Features

Adaptive modulation and coding (AMC), Hybrid Automatic Repeat Request (HARQ) and Fast Channel Feedback (CQICH) were introduced with Mobile WiMAX to enhance coverage and capacity for WiMAX in mobile applications.

Support for QPSK, 16QAM and 64QAM are mandatory in the DL with Mobile WiMAX. In the UL, 64QAM is optional. Both Convolutional Code (CC) and Convolutional Turbo Code (CTC) with variable code rate and repetition coding are supported. Block Turbo Code and Low Density Parity Check Code (LDPC) are supported as optional features. Table 2 summarizes the coding and modulation schemes supported in the Mobile WiMAX profile the optional UL codes and modulation are shown in italics.

		DL	UL
Modulation		QPSK, 16QAM, 64QAM	QPSK,16QAM, *64QAM*
Code Rate	CC	1/2, 2/3, 3/4	1/2, 2/3, 3/4
	CTC	1/2, 2/3, 3/4, 5/6	1/2, 2/3, *5/6*
	Repetition	x2, x4, x6	x2, x4, x6

Table 2: Supported Code and Modulations

The combinations of various modulations and code rates provide a fine resolution of data rates as shown in Table 3 which shows the data rates for 5 and 10 MHz channels with PUSC sub-channels. The frame duration is 5 milliseconds. Each frame has 48 OFDM symbols, with 44 OFDM symbols available for data transmission. The highlighted values indicate data rates for optional 64QAM in the UL.

Parameter	Downlink	Uplink	Downlink	Uplink
System Bandwidth	5 MHz		10 MHz	
FFT Size	512		1024	
Null Sub-Carriers	92	104	184	184
Pilot Sub-Carriers	60	136	120	280
Data Sub-Carriers	360	272	720	560
Sub-Channels	15	17	30	35

Symbol Period, T_S	102.9 microseconds
Frame Duration	5 milliseconds
OFDM Symbols/Frame	48
Data OFDM Symbols	44

Mod.	Code Rate	5 MHz Channel Downlink Rate, Mbps	5 MHz Channel Uplink Rate, Mbps	10 MHz Channel Downlink Rate, Mbps	10 MHz Channel Uplink Rate, Mbps
QPSK	1/2 CTC, 6x	0.53	0.38	1.06	0.78
	1/2 CTC, 4x	0.79	0.57	1.58	1.18
	1/2 CTC, 2x	1.58	1.14	3.17	2.35
	1/2 CTC, 1x	3.17	2.28	6.34	4.70
	3/4 CTC	4.75	3.43	9.50	7.06
16QAM	1/2 CTC	6.34	4.57	12.07	9.41
	3/4 CTC	9.50	6.85	19.01	14.11
64QAM	1/2 CTC	9.50	6.85	19.01	14.11
	2/3 CTC	12.67	9.14	26.34	18.82
	3/4 CTC	14.26	10.28	28.51	21.17
	5/6 CTC	15.84	11.42	31.68	23.52

Table 3: Mobile WiMAX PHY Data Rates with PUSC Sub-Channel[1]

The base station scheduler determines the appropriate data rate (or burst profile) for each burst allocation based on the buffer size, channel propagation conditions at the receiver, etc. A Channel Quality Indicator (CQI) channel is utilized to provide channel-state information from the user terminals to the base station scheduler. Relevant channel-state information can be fed back by the CQICH including: Physical CINR, effective CINR, MIMO mode selection and frequency selective sub-channel selection. With TDD implementations, link adaptation can also take advantage of channel reciprocity to provide a more accurate measure of the channel condition (such as sounding).

Hybrid Auto Repeat Request (HARQ) is supported by Mobile WiMAX. HARQ is enabled using *N* channel "Stop and Wait" protocol which provides fast response to packet errors and improves cell edge coverage. Chase Combining and optionally, Incremental Redundancy are supported to further improve the reliability of the retransmission. A dedicated ACK channel is also provided in the uplink for HARQ ACK/NACK signaling. Multi-channel HARQ operation is supported. Multi-channel stop-and-wait ARQ with a small number of channels is an efficient, simple protocol that minimizes the memory required for HARQ and stalling. WiMAX provides signaling to allow fully asynchronous operation. The asynchronous operation allows variable delay between retransmissions which gives more flexibility to the scheduler at the cost of additional overhead for each retransmission allocation. HARQ combined together with CQICH and AMC provides robust link adaptation in mobile environments at vehicular speeds in excess of 120 km/hr.

2.6 MAC Layer Description

The 802.16 standard was developed from the outset for the delivery of broadband services including voice, data, and video. The MAC layer is based on the time-proven DOCSIS standard and can support bursty data traffic with high peak rate demand while simultaneously supporting streaming video and latency-sensitive voice traffic over the same channel. The resource allocated to one terminal by the MAC scheduler can vary from a single time slot to the entire frame, thus providing a very large dynamic range of throughput to a specific user terminal at any given time. Furthermore, since the resource allocation information is conveyed in the MAP messages at the beginning of each frame, the scheduler can effectively change the resource allocation on a frame-by-frame basis to adapt to the bursty nature of the traffic.

2.6.1 Quality of Service (QoS) Support

With fast air link, symmetric downlink/uplink capacity, fine resource granularity and a flexible resource allocation mechanism, Mobile WiMAX can meet QoS requirements for a wide range of data services and applications.

In the Mobile WiMAX MAC layer, QoS is provided via service flows as illustrated in Figure 8. This is a unidirectional flow of packets that is provided with a particular set of QoS parameters. Before providing a certain

[1] PHY Data Rate=(Data sub-carriers/Symbol period)*(information bits per symbol)

type of data service, the base station and user-terminal first establish a unidirectional logical link between the peer MACs called a connection. The outbound MAC then associates packets traversing the MAC interface into a service flow to be delivered over the connection. The QoS parameters associated with the service flow define the transmission ordering and scheduling on the air interface. The connection-oriented QoS therefore, can provide accurate control over the air interface. Since the air interface is usually the bottleneck, the connection-oriented QoS can effectively enable the end-to-end QoS control. The service flow parameters can be dynamically managed through MAC messages to accommodate the dynamic service demand. The service flow based QoS mechanism applies to both DL and UL to provide improved QoS in both directions. Mobile WiMAX supports a wide range of data services and applications with varied QoS requirements. These are summarized in Table 4.

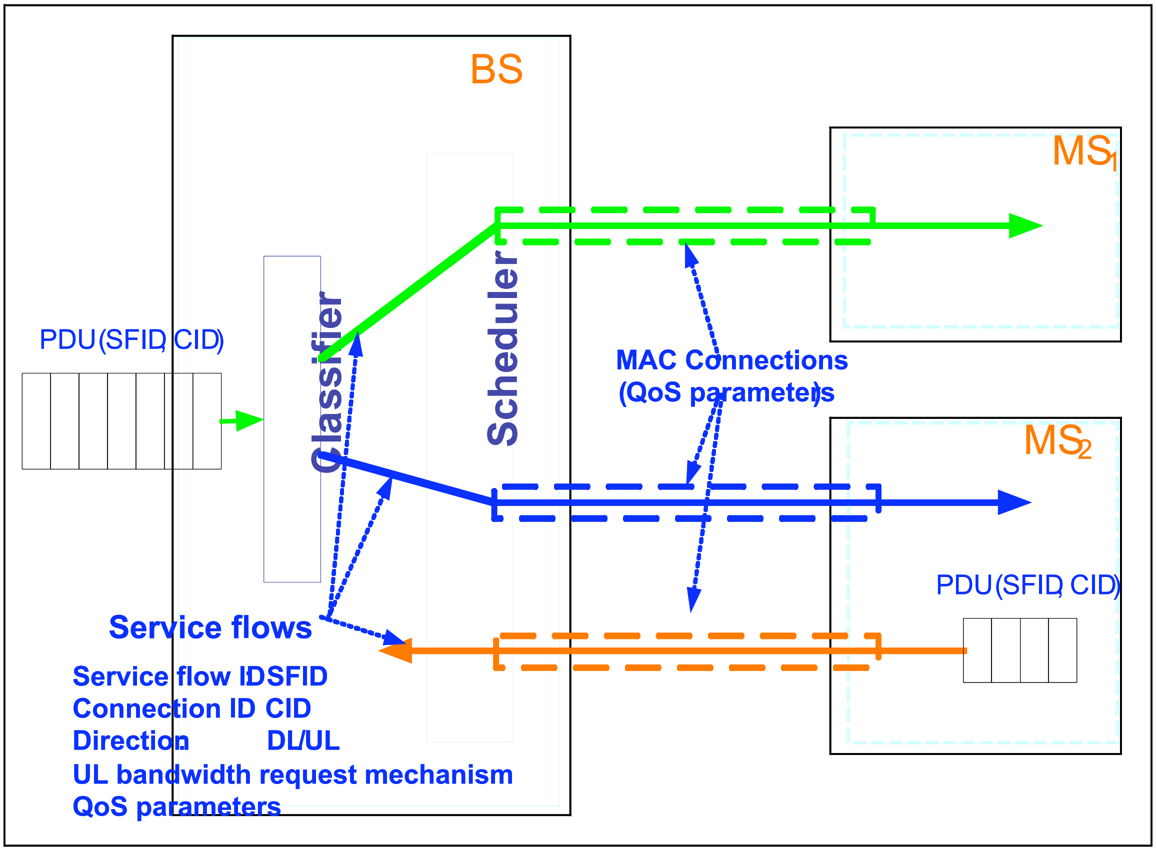

Figure 10: Mobile WiMAX QoS Support

QoS Category	Applications	QoS Specifications
UGS Unsolicited Grant Service	VoIP	• Maximum Sustained Rate • Maximum Latency Tolerance • Jitter Tolerance
rtPS Real-Time Packet Service	Streaming Audio or Video	• Minimum Reserved Rate • Maximum Sustained Rate • Maximum Latency Tolerance • Traffic Priority
ErtPS Extended Real-Time Packet Service	Voice with Activity Detection (VoIP)	• Minimum Reserved Rate • Maximum Sustained Rate • Maximum Latency Tolerance • Jitter Tolerance • Traffic Priority

QoS Category	Applications	QoS Specifications
nrtPS Non-Real-Time Packet Service	File Transfer Protocol (FTP)	• Minimum Reserved Rate • Maximum Sustained Rate • Traffic Priority
BE Best-Effort Service	Data Transfer, Web Browsing, etc.	• Maximum Sustained Rate • Traffic Priority

Table 4: Mobile WiMAX Applications and Quality of Service

2.6.2 MAC Scheduling Service

The Mobile WiMAX MAC scheduling service is designed to efficiently deliver broadband data services including voice, data, and video over time varying broadband wireless channel. The MAC scheduling service has the following properties that enable the broadband data service:

- **Fast Data Scheduler:** The MAC scheduler must efficiently allocate available resources in response to bursty data traffic and time-varying channel conditions. The scheduler is located at each base station to enable rapid response to traffic requirements and channel conditions. The data packets are associated to service flows with well defined QoS parameters in the MAC layer so that the scheduler can correctly determine the packet transmission ordering over the air interface. The CQICH channel provides fast channel information feedback to enable the scheduler to choose the appropriate coding and modulation for each allocation. The adaptive modulation/coding combined with HARQ provide robust transmission over the time-varying channel.
- **Scheduling for both DL and UL:** The scheduling service is provided for both DL and UL traffic. In order for the MAC scheduler to make an efficient resource allocation and provide the desired QoS in the UL, the UL must feedback accurate and timely information as to the traffic conditions and QoS requirements. Multiple uplink bandwidth request mechanisms, such as bandwidth request through ranging channel, piggyback request and polling are designed to support UL bandwidth requests. The UL service flow defines the feedback mechanism for each uplink connection to ensure predictable UL scheduler behavior. Furthermore, with orthogonal UL sub-channels, there is no intra-cell interference. UL scheduling can allocate resource more efficiently and better enforce QoS.
- **Dynamic Resource Allocation**: The MAC supports frequency-time resource allocation in both DL and UL on a per-frame basis. The resource allocation is delivered in MAP messages at the beginning of each frame. Therefore, the resource allocation can be changed on frame-by-frame in response to traffic and channel conditions. Additionally, the amount of resource in each allocation can range from one slot to the entire frame. The fast and fine granular resource allocation allows superior QoS for data traffic.
- **QoS Oriented:** The MAC scheduler handles data transport on a connection-by-connection basis. Each connection is associated with a single data service with a set of QoS parameters that quantify the aspects of its behavior. With the ability to dynamically allocate resources in both DL and UL, the scheduler can provide superior QoS for both DL and UL traffic. Particularly with uplink scheduling - *the uplink resource is more efficiently allocated, performance is more predictable, and QoS is better enforced.*
- **Frequency Selective Scheduling:** The scheduler can operate on different types of sub-channels. For frequency-diverse sub-channels such as PUSC permutation, where sub-carriers in the sub-channels are pseudo-randomly distributed across the bandwidth, sub-channels are of similar quality. Frequency-diversity scheduling can support a QoS with fine granularity and flexible time-frequency resource scheduling. With contiguous permutation such as AMC permutation, the sub-channels may experience different attenuation. The frequency-selective scheduling can allocate mobile users to their corresponding strongest sub-channels. The frequency-selective scheduling can enhance system capacity with a moderate increase in CQI overhead in the UL.

2.6.3 Mobility Management

Battery life and handoff are two critical issues for mobile applications. Mobile WiMAX supports Sleep Mode and Idle Mode to enable power-efficient MS operation. Mobile WiMAX also supports seamless handoff to enable the MS to switch from one base station to another at vehicular speeds without interrupting the connection.

2.6.4 Power Management

Mobile WiMAX supports two modes for power efficient operation – Sleep Mode and Idle Mode. Sleep Mode is a state in which the MS conducts pre-negotiated periods of absence from the Serving Base Station air interface. These periods are characterized by the unavailability of the MS, as observed from the Serving Base Station, to DL or UL traffic. Sleep Mode is intended to minimize MS power usage and minimize the usage of the Serving Base Station air interface resources. The Sleep Mode also provides flexibility for the MS to scan other base stations to collect information to assist handoff during the Sleep Mode.

Idle Mode provides a mechanism for the MS to become periodically available for DL broadcast traffic messaging without registration at a specific base station as the MS traverses an air link environment populated by multiple base stations. Idle Mode benefits the MS by removing the requirement for handoff and other normal operations and benefits the network and base station by eliminating air interface and network handoff traffic from essentially inactive MSs while still providing a simple and timely method (paging) for alerting the MS about pending DL traffic.

2.6.5 Handoff

There are three handoff methods supported within the 802.16e standard – Hard Handoff (HHO), Fast Base Station Switching (FBSS) and Macro Diversity Handover (MDHO). Of these, the HHO is mandatory while FBSS and MDHO are two optional modes. The WiMAX Forum has developed several techniques for optimizing hard handoff within the framework of the 802.16e standard. These improvements have been developed with the goal of keeping Layer 2 handoff delays to less than 50 milliseconds.

When FBSS is supported, the MS and BS maintain a list of BSs that are involved in FBSS with the MS. This set is called an Active Set. In FBSS, the MS continuously monitors the base stations in the Active Set. Among the BSs in the Active Set, an Anchor BS is defined. When operating in FBSS, the MS only communicates with the Anchor BS for uplink and downlink messages including management and traffic connections. Transition from one Anchor BS to another (i.e. BS switching) is performed without invocation of explicit HO signaling messages. Anchor update procedures are enabled by communicating signal strength of the serving BS via the CQI channel. A FBSS handover begins with a decision by an MS to receive or transmit data from the Anchor BS that may change within the active set. The MS scans the neighbor BSs and selects those that are suitable to be included in the active set. The MS reports the selected BSs and the active set update procedure is performed by the BS and MS. The MS continuously monitors the signal strength of the BSs that are in the active set and selects one BS from the set to be the Anchor BS. The MS reports the selected Anchor BS on CQICH or MS initiated HO request message. An important requirement of FBSS is that the data is simultaneously transmitted to all members of an active set of BSs that are able to serve the MS.

For MSs and BSs that support MDHO, the MS and BS maintain an active set of BSs that are involved in MDHO with the MS. Among the BSs in the active set, an Anchor BS is defined. The regular mode of operation refers to a particular case of MDHO with the active set consisting of a single BS. When operating in MDHO, the MS communicates with all BSs in the active set of uplink and downlink unicast messages and traffic. A MDHO begins when a MS decides to transmit or receive unicast messages and traffic from multiple BSs in the same time interval. For downlink MDHO, two or more BSs provide synchronized transmission of MS downlink data such that diversity combining is performed at the MS. For uplink MDHO, the transmission from a MS is received by multiple BSs where selection diversity of the information received is performed.

2.6.6 Security

Mobile WiMAX supports best in class security features by adopting the best technologies available today. Support exists for mutual device/user authentication, flexible key management protocol, strong traffic encryption, control and management plane message protection and security protocol optimizations for fast handovers.

The usage aspects of the security features are:

- **Key Management Protocol:** Privacy and Key Management Protocol Version 2 (PKMv2) is the basis of Mobile WiMAX security as defined in 802.16e. This protocol manages the MAC security using PKM-REQ/RSP messages. PKM EAP authentication, Traffic Encryption Control, Handover Key Exchange and Multicast/Broadcast security messages all are based on this protocol.
- **Device/User Authentication:** Mobile WiMAX supports Device and User Authentication using IETF EAP protocol by providing support for credentials that are SIM-based, USIM-based or Digital Certificate or

UserName/Password-based. Corresponding EAP-SIM, EAP-AKA, EAP-TLS or EAP-MSCHAPv2 authentication methods are supported through the EAP protocol. Key deriving methods are the only EAP methods supported.

- **Traffic Encryption:** AES-CCM is the cipher used for protecting all the user data over the Mobile WiMAX MAC interface. The keys used for driving the cipher are generated from the EAP authentication. A Traffic Encryption State machine that has a periodic key (TEK) refresh mechanism enables sustained transition of keys to further improve protection.
- **Control Message Protection:** Control data is protected using AES based CMAC, or MD5-based HMAC schemes.
- **Fast Handover Support:** A 3-way Handshake scheme is supported by Mobile WiMAX to optimize the re-authentication mechanisms for supporting fast handovers. This mechanism is also useful to prevent any man-in-the-middle-attacks.

2.7 References

[1] IEEE Std 802.16e-2005, Amendment for Combined Fixed and Mobile Broadband Wireless Access Systems, December 2005

[2] IEEE Std 802.16™-2004, IEEE Standard for Local and metropolitan area networks: Part 16: Air Interface for Fixed Broadband Wireless Access Systems, June 2004

[3] Mobile WiMAX System Profile

[4] Mobile WiMAX – Part I: A Technical Overview and Performance Evaluation, August 2006, WiMAX Forum

[5] Mobile WiMAX Network Architecture and Specifications

Annex A. Radio Transmission Technology (RTT) Description Template

Attachment 1

TABLE CONTENTS

A1.1	Test environment support	
A1.1.1	In what test environments will the RTT operate?	- indoor - outdoor to indoor and pedestrian, - vehicular - mixed
A1.1.2	If the RTT supports more than one test environment, what test environment does this technology description template address?	One template for all
A1.1.3	Does the RTT include any features in support of FWA application? Provide detail about the impact of those features on the technical parameters provided in this template, stating whether the technical parameters provided apply for mobile as well as for FWA applications.	Yes. Flexible mixed fixed and mobile design. - QoS - Dynamic bandwidth allocation - Continuous and variable bit rate support - Support of nomadic operation - Support of fixed wireless voice and data services - Etc. Yes, see Recommendation ITU-R F.1763
A1.2	Technical parameters NOTE 1 – Parameters for both forward link and reverse link should be described separately, if necessary.	
A1.2.1	What is the minimum frequency band required to deploy the system (MHz)?	10 MHz. Even though system can function in 5MHz, 10 MHz is recommended for optimal performance.
A1.2.2	What is the duplex method: TDD or FDD?	TDD
A1.2.2.1	What is the minimum up/down frequency separation for FDD?	N/A
A1.2.2.2	What is requirement of transmit/receive isolation? Does the proposal require a duplexer in either the mobile station (MS) or BS?	Does not require a duplexer.
A1.2.3	Does the RTT allow asymmetric transmission to use the available spectrum? Characterize.	Yes. The ratio of uplink to downlink transmission can be reconfigured on a system-wide basis.
A1.2.4	What is the RF channel spacing (kHz)? In addition, does the RTT use an interleaved frequency plan? 10000 KHz NOTE 1 – The use of the second adjacent channel instead of the adjacent channel at a neighbouring cluster cell is called "interleaved frequency planning". If a proponent is going to employ an interleaved frequency plan, the proponent should state so in § A1.2.4 and complete § A1.2.15 with the protection ratio for both the adjacent and second adjacent channel.	10000 KHz The RTT does not use an interleaved frequency plan
A1.2.5	What is the bandwidth per duplex RF channel (MHz) measured at the 3 dB down points? It is given by (bandwidth per RF channel) × (1 for TDD and 2 for FDD). Provide detail.	Nominally 10 MHz (TDD). Measured at the 3dB down points is roughly about 9.4MHz, depending on the permutation used.

A1.2.5.1	Does the proposal offer multiple or variable RF channel bandwidth capability? If so, are multiple bandwidths or variable bandwidths provided for the purposes of compensating the transmission medium for impairments but intended to be feature transparent to the end user?	The RTT offers variable RF channel bandwidth capability through the use of OFDMA subchannelization.
A1.2.6	What is the RF channel bit rate (kbit/s)? NOTE 1 – The maximum modulation rate of RF (after channel encoding, adding of in-band control signalling and any overhead signalling) possible to transmit carrier over an RF channel, i.e. independent of access technology and of modulation schemes.	**DOWNLINK** **Distributed permutation of subcarriers** Assumptions: 32 data symbols per frame (35 symbols in subframe, 1 symbol for preamble, 2 symbols for control information), 5msec frame duration, 64QAM 5/6 code rate, 30 slots for 2 symbols, 48 data tones per slot. Maximum data rate: 23040kbit/s **UPLINK** **Distributed permutation of subcarriers** Assumptions: 18 data symbols per frame (21 symbols in UL subframe, 3 symbols for control channels), 5msec frame duration, 16QAM 3/4 code rate, 35 slots for 3 symbols, 48 data tones per slot. Maximum data rate: 6048kbit/s

A1.2.7	*Frame structure*: describe the frame structure to give sufficient information such as: – frame length, – the number of time slots per frame, – guard time or the number of guard bits, – user information bit rate for each time slot, – channel bit rate (after channel coding), – channel symbol rate (after modulation), – associated control channel (ACCH) bit rate, – power control bit rate. NOTE 1 – Channel coding may include forward error correction (FEC), cyclic redundancy checking (CRC), ACCH, power control bits and guard bits. Provide detail. NOTE 2 – Describe the frame structure for forward link and reverse link, respectively. NOTE 3 – Describe the frame structure for each user information rate.	Frame length : 5ms The number of time slots per frame : N/A The number of time symbols per frame : 47 symbols The number of subcarriers per each symbol : 1024 FFT Resource allocation : 2 dimensional structure for frequency and time (see section 2.4 of the RTT System Description for more details) Subchannel structure : see section 2.2 of the RTT System Description for details Ratio of DL and UL subframe : Ranging from 35 symbols:12 symbols to 26 symbols:21 symbols (DL:UL) TTG / RTG : 105.7 _sec / 60 _sec Common control overhead : 1 symbol per frame for preamble (see section 2.4 of the RTT System Description for more details) **DOWNLINK** (See A1.2.5.1) **Distributed permutation of subcarriers** The number of subcarriers per slot : 48 (data) + 8 (pilots) Guard subcarrier : 184 (including DC subcarrier) The channel bit or symbol rate is variable, depending on the number of allocated slots, and the modulation and coding rate. Power control rate : no power control **Adjacent permutation of subcarriers** The number of subcarriers per slot : 48 (data) + 6 (pilots) Guard subcarrier : 160 (including DC subcarrier) **UPLINK** **Distributed permutation of subcarriers** The number of subcarriers per slot : 48 (data) + 24 (pilots) Guard subcarrier : 184 (including DC subcarrier)

A1.2.8	Does the RTT use frequency hopping? If so, characterize and explain particularly the impact (e.g. improvements) on system performance.	No
A1.2.8.1	What is the hopping rate?	N/A
A1.2.8.2	What is the number of the hopping frequency sets?	N/A
A1.2.8.3	Are BSs synchronized or non-synchronized?	N/A
A1.2.9	Does the RTT use a spreading scheme?	No
A1.2.9.1	What is the chip rate (Mchip/s)? Rate at input to modulator.	N/A
A1.2.9.2	What is the processing gain? 10 log (chip rate/information rate).	N/A
A1.2.9.3	Explain the uplink and downlink code structures and provide the details about the types (e.g. personal numbering (PN) code, Walsh code) and purposes (e.g. spreading, identification, etc.) of the codes.	N/A
A1.2.10	Which access technology does the proposal use: TDMA, FDMA, CDMA, hybrid, or a new technology? In the case of CDMA, which type of CDMA is used: frequency hopping (FH) or direct sequence (DS) or hybrid? Characterize.	OFDMA
A1.2.11	What is the baseband modulation technique? If both the data modulation and spreading modulation are required, describe in detail. What is the peak to average power ratio after baseband filtering (dB)?	**DOWNLINK** QPSK, 16 QAM, 64 QAM for data modulation. Spreading modulation does not apply. **UPLINK** QPSK, 16 QAM for data modulation. Spreading modulation does not apply. PAPR is about 12 dB without any PAPR reduction scheme.
A1.2.12	What are the channel coding (error handling) rate and form for both the forward and reverse links? E.g., does the RTT adopt: – FEC or other schemes? – Unequal error protection? Provide details. – Soft decision decoding or hard decision decoding? Provide details. – Iterative decoding (e.g. turbo codes)? Provide details. – Other schemes?	Convolutional Coding is mandatory and Convolutional Turbo Coding is also supported Modulation schemes: QPSK, 16QAM and 64QAM for downlink, QPSK and 16QAM for uplink. Coding rates: QPSK 1/2, QPSK 3/4, 16QAM 1/2, 16QAM 3/4, 64QAM 1/2, 64QAM 2/3, 64QAM 3/4, 64QAM 5/6. Coding repetition rates: 1x, 2x, 4x and 6x. Unequal error protection : None Soft decision decoding and iterative decoding: It is an implementation issue not covered by the description.
A1.2.13	What is the bit interleaving scheme? Provide detailed description for both uplink and downlink.	The bit interleaving scheme is the same for both uplink and downlink. All encoded data bits shall be interleaved by a block interleaver with a block size corresponding to the number of coded bits per the encoded block size.

A1.2.14	Describe the approach taken for the receives (MS and BS) to cope with multipath propagation effects (e.g. via equalizer, Rake receiver, etc.).	To cope with the multipath propagation effect, the cyclic prefix and 1-tap equalizer are employed. The length of cyclic prefix is 1/8 of symbol duration thus 11.4 _sec.
A1.2.14.1	Describe the robustness to intersymbol interference and the specific delay spread profiles that are best or worst for the proposal.	The intersymbol interference can be removed by the use of sufficiently longer cyclic prefix than delay spread.
A1.2.14.2	Can rapidly changing delay spread profile be accommodated? Describe.	Yes, delay spread variation within the length of cyclic prefix does not cause the intersymbol interference.
A1.2.15	What is the adjacent channel protection ratio? NOTE 1 – In order to maintain robustness to adjacent channel interference, the RTT should have some receiver characteristics that can withstand higher power adjacent channel interference. Specify the maximum allowed relative level of adjacent RF channel power (dBc). Provide detail how this figure is assumed.	Min adjacent channel rejection at BER=10-6 for 3 dB degradation C/I 11 dB - 16QAM, 3/4 coding rate 4 dB - 64 QAM, 2/3 coding rate Min non-adjacent channel rejection at BER=10-6 for 3 dB degradation C/I 30 dB - 16QAM, 3/4 coding rate 23 dB - 64 QAM, 2/3 coding rate
A1.2.16	Power classes	Transmit power (dBm) for 16QAM 1. 18 <= Ptx,max < 21 2. 21 <= Ptx,max < 25 3. 25 <= Ptx,max < 30 4. 30 <= Ptx,max Transmit power (dBm) for QPSK 1. 20 <= Ptx,max < 23 2. 23 <= Ptx,max < 27 3. 27 <= Ptx,max < 30 4. 30 <= Ptx,max
A1.2.16.1	*Mobile terminal emitted power*: what is the radiated antenna power measured at the antenna? For terrestrial component, give (dBm). For satellite component, the mobile terminal emitted power should be given in e.i.r.p. (effective isotropic radiated power) (dBm).	See A.1.2.16
A1.2.16.1.1	What is the maximum peak power transmitted while in active or busy state?	See A.1.2.16
A1.2.16.1.2	What is the time average power transmitted while in active or busy state? Provide detailed explanation used to calculate this time average power.	See A.1.2.16
A1.2.16.2	Base station transmit power per RF carrier for terrestrial component	See A.1.2.16
A1.2.16.2.1	What is the maximum peak transmitted power per RF carrier radiated from antenna?	Not limited by RTT
A1.2.16.2.2	What is the average transmitted power per RF carrier radiated from antenna?	Not limited by RTT

A1.2.17	What is the maximum number of voice channels available per RF channel that can be supported at one BS with 1 RF channel (TDD systems) or 1 duplex RF channel pair (FDD systems), while still meeting ITU-T Recommendation G.726 performance requirements?	The maximum number of voice channels per 1 RF channel depends on the bit rate and sampling rate supported by the codecs defined in the G.726. For instance, in case of the bit rate of 16kbps with 20msec sampling rate, up to 256 users can be supported simultaneously by a RF channel.
A1.2.18	*Variable bit rate capabilities*: describe the ways the proposal is able to handle variable baseband transmission rates. For example, does the RTT use: – adaptive source and channel coding as a function of RF signal quality? – Variable data rate as a function of user application? – Variable voice/data channel utilization as a function of traffic mix requirements? Characterize how the bit rate modification is performed. In addition, what are the advantages of your system proposal associated with variable bit rate capabilities?	Variable bit rate is supported by the flexible resource allocation. By assigning the variable number of subchannels and using various modulations and coding rates frame by frame, bit rate can be variable frame by frame. Modulation and coding rate is usually defined by user's RF signal quality (CQI). For higher data rates, the bit rate information is provided to the receiver via scheduling mechanisms and associated control signalling every frame.
A1.2.18.1	What are the user information bit rates in each variable bit rate mode?	The user information bit rates are variable according to the number of subachannels assigned and modulation and coding rate used. **DOWNLINK** Modulation : QPSK, 16QAM, 64QAM Coding rate : 1/2, 2/3, 3/4, 5/6 3312 kbps(1/2, QPSK, (DL:UL)=(26:21) symbols) ~ 23040 kbps(5/6, 64QAM, (DL:UL)=(35:12) symbols)* **UPLINK** Modulation : QPSK, 16QAM Coding rate : 1/2, 3/4 1008 kbps(1/2, QPSK, (DL:UL)=(35:12) symbols) ~ 6048 kbps(3/4, 16QAM, (DL:UL)=(26:21) symbols)* *PHY Data Rate=(Data sub-carriers/Symbol period)*(information bits per symbol) [reference by WiMAX white paper]

A1.2.19	What kind of voice coding scheme or codec is assumed to be used in proposed RTT? If the existing specific voice coding scheme or codec is to be used, give the name of it. If a special voice coding scheme or codec (e.g. those not standardized in standardization bodies such as ITU) is indispensable for the proposed RTT, provide detail, e.g. scheme, algorithm, coding rates, coding delays and the number of stochastic code books.	Due to the IP- based characteristics of the radio interface it can utilize any speech codec.
A1.2.19.1	Does the proposal offer multiple voice coding rate capability? Provide detail.	Yes. The RTT supports flexible data rate for each users and also provide variety scheduling services. A constant bit rate is provided by UGS service, while a variable bit rate is provided by ErtPS service.. See A.1.2.18, A1.2.20.1 and A1.2.20.2
A1.2.20	*Data services* : are there particular aspects of the proposed technologies which are applicable for the provision of circuit-switched, packet-switched or other data services like asymmetric data services? For each service class (A, B, C and D) a description of RTT services should be provided, at least in terms of bit rate, delay and BER/frame error rate (FER). NOTE 1 – See Recommendation ITU-R M.1224 for the definition of: – "circuit transfer mode", – "packet transfer mode", – "connectionless service", and for the aid of understanding "circuit switched" and "packet switched" data services. NOTE 2 – See ITU-T Recommendation I.362 for details about the service classes A, B, C and D.	Yes, a wide range of data services and application with varied QoS requirements are supported. These are summarized together with guidelines for bit rate, latency, traffic priority and jitter to assure a quality user experience. And it enables flexible support of simultaneous use of a diverse set of IP services.
A1.2.20.1	For delay constrained, connection oriented (Class A).	The RTT provides UGS (unsolicited grant service), corresponding to the Class A. UGS is characterized as constant and low data rates and low delay data service. Data rates is ranged from 32 Kbps to 64 Kbps and latency is required to be less than 160 msec.

A1.2.20.2	For delay constrained, connection oriented, variable bit rate (Class B).	The RTT provides rtPS (real-time polling service), corresponding to the Class B. rtPS is characterized as low to high data rates. Data rates is ranged from 5 Kbps to 2 Mbps. The PTT provides ErtPS (extended real-time polling service) as well. ErtPS is characterized as low data rates and low delay data service. Data rates is fixed to 50 Kbps and latency is required to be less than 25 msec.
A1.2.20.3	For delay unconstrained, connection oriented (Class C).	The RTT provides nrtPS (non-real-time polling service), corresponding to the Class C. nrtPS is characterized as high data rates service. Data rates is required to be more than 2 Mbps.
A1.2.20.4	For delay unconstrained, connectionless (Class D).	The RTT provides BE (best effort service) corresponding to the Class D. BE is characterized as moderate data rates service. Data rates is ranged from 10 Kbps to 2 Mbps.
A1.2.21	Simultaneous voice/data services: is the proposal capable of providing multiple user services simultaneously with appropriate channel capacity assignment? NOTE 1 – The following describes the different techniques that are inherent or improve to a great extent the technology described above to be presented. Description for both BS and MS are required in attributes from § A1.2.22 through § A1.2.23.2.	Yes, multiple parallel services are supported with different QoS requirements. Each service is associated with a set of QoS parameters that quantify aspects of its behavior. These parameters are managed using the dynamic service provisions, represented by the DSA and DSC message dialog.
A1.2.22	*Power control characteristics*: is a power control scheme included in the proposal? Characterize the impact (e.g. improvements) of supported power control schemes on system performance.	Yes. A closed loop power control scheme and an open loop power control scheme are included. By mean of these power control schemes, the interference level is reduced and the uplink system level throughput is increased.

A1.2.22.1	What is the power control step size (dB)?	Power control step size is variable ranging from 0.25 dB to 32 dB. An 8-bit signed integer in power control information element indicates the power control step size in 0.25 dB units.
A1.2.22.2	What are the number of power control cycles per second?	The power control cycle of closed-loop power control is dependent on the rate of power control information element transmission, but less than 200 Hz. Due to TDD nature, the open loop power control cycle is inherently identical to the number of frames per seconds, thus 200 Hz.
A1.2.22.3	What is the power control dynamic range (dB)?	The minimum power control dynamic range is 45dB.
A1.2.22.4	What is the minimum transmit power level with power control?	The RTT supports 45 dB under the full power assumption
A1.2.22.5	What is the residual power variation after power control when RTT is operating? Provide details about the circumstances (e.g. in terms of system characteristics, environment, deployment, MS-speed, etc.) under which this residual power variation appears and which impact it has on the system performance.	The accuracy for power level control can vary from +/-0.5 dB to +/-2dB depending on the power control step size. +/- 0.5 dB for step size +/-1 dB +/- 1.0 dB for step size +/-2 dB +/- 1.5 dB for step size +/-3 dB +/- 2.0 dB for otherwise
A1.2.23	*Diversity combining in MS and BS*: are diversity combining schemes incorporated in the design of the RTT?	Yes.

A1.2.23.1	Describe the diversity techniques applied in the MS and at the BS, including micro diversity and macro diversity, characterizing the type of diversity used, for example: – time diversity: repetition, Rake-receiver, etc., – space diversity: multiple sectors, multiple satellite, etc., – frequency diversity: FH, wideband transmission, etc., – code diversity: multiple PN codes, multiple FH code, etc., – other scheme. Characterize the diversity combining algorithm, for example, switch diversity, maximal ratio combining, equal gain combining. Additionally, provide supporting values for the number of receivers (or demodulators) per cell per mobile user. State the dB of performance improvement introduced by the use of diversity. For the MS: what is the minimum number of RF receivers (or demodulators) per mobile unit and what is the minimum number of antennas per mobile unit required for the purpose of diversity reception? These numbers should be consistent to that assumed in the link budget template of Annex 2 and that assumed in the calculation of the “capacity” defined at § A1.3.1.5.	The following diversity schemes are provided. Time diversity : interleaving and FEC Space diversity : multiple antennas (Space-time block coding transmission scheme, cyclic delay diversity, receiver diversity) 1 antenna, 2 antennas (optional); The number of BS transmission antenna is 1 while the number of BS receive antennas is up to 4. The number of MS transmission antenna is 1 while the number of MS receive antennas is up to 2. Diversity combining : maximal-ratio combining or MMSE may be used with multiple antennas Frequency diversity : distributed permutation over wideband Minimum number of RF receivers : 1 per MS Minimum number of antennas per mobile unit : 1 per MS
A1.2.23.2	What is the degree of improvement expected (dB)? Also indicate the assumed conditions such as BER and FER.	(See reference [4]) Cyclic delay diversity combining is expected to achieve 3 dB gain. Rx antenna diversity is expected to achieve 3 dB gain (BS and MS with 2 antennas, respectively). With 2x2 MIMO, the spectral efficiency is further improved by 55% to 60% in the downlink and by about 35% in the uplink.

A1.2.24	*Handover/automatic radio link transfer (ALT)* : do the radio transmission technologies support handover? Yes Characterize the type of handover strategy (or strategies) which may be supported, e.g. MS assisted handover. Give explanations on potential advantages, e.g. possible choice of handover algorithms. Provide evidence whenever possible.	[See [1] 6.3.22 MAC layer handover procedures] Yes. The RTT supports handover and also provides means for expediting handover. Each base station broadcasts the information on the list of neighboring base stations and their channel information such as the operating center frequency, preamble index and synchronization periodically. The channel information in this broadcasting is used for a mobile station to synchronize with the neighboring base station. After a mobile station monitors the signal strength of a neighboring base station and seeks suitable base station(s) for handover, the mobile station or its serving base station can initiate handover by handover request message. But only the mobile station can transmit handover indication message to the its serving base station. After transmitting handover indication message, the mobile station stops monitoring the downlink frame of its serving base station and performs network re-entry to target base station. To reduce the handover latency further, the serving base station provides the target base station with network entry information on a mobile station to be handed over the target base station.
A1.2.24.1	What is the break duration (s) when a handover is executed? In this evaluation, a detailed description of the impact of the handover on the service performance should also be given. Explain how the estimate was derived.	See Annex C for definitions and details
A1.2.24.2	For the proposed RTT, can handover cope with rapid decrease in signal strength (e.g. street corner effect)? Give a detailed description of: – the way the handover detected, initiated and executed, – how long each of this action lasts (minimum/maximum time (ms)), – the time-out periods for these actions.	[See [1] 11.1.7 MOB-NBR-ADV message encodings] A base station broadcasts the criterion which is being used for mobile station to request handover. The mobile station issues handover request message whenever the criterion is met. Design of criterion depends on the implementation but usually the received signal strength by a mobile station is used.

A1.2.25	Characterize how the proposed RTT reacts to the system deployment (e.g. necessity to add new cells and/or new carriers) particularly in terms of frequency planning.	All base stations can use the same frequency or different frequency depending on the frequency reuse deployment scenario. OFDMA subchannelization allows various permutations of subcarriers. A distributed permutation of subcarriers, e.g., PUSC (partial usage of sub-carrier) in this RTT, minimizes interferences from neighboring cells and/or sectors in case of the frequency reuse of 1. Different operators usually use different frequencies.
A1.2.26	*Sharing frequency band capabilities*: to what degree is the proposal able to deal with spectrum sharing among IMT-2000 systems as well as with all other systems: – spectrum sharing between operators, – spectrum sharing between terrestrial and satellite IMT-2000 systems, – spectrum sharing between IMT-2000 and non-IMT-2000 systems, – other sharing schemes.	The proposed RTT utilizes scalable OFDMA which has inherent interference protection capabilities due to allocation of a varying subset of available sub-carriers to different users. This capability, complemented by interference mitigation techniques described in ITU-R Report M.2045 such as use of appropriate filters and linear power amplifiers would ensure excellent potential for optimum spectrum sharing between the proposed RTT and other IMT-2000 systems. ITU-R WP 8F is in the process of performing sharing studies between fixed/nomadic and mobile IEEE 802.16 and IMT-2000. Preliminary results show similarities with the case of coexistence between IMT-2000 TDD and FDD technologies as captured in Reports ITU-R M.2030 and M.2045. The RF parameters appropriate for use in sharing studies for the above studies have been provided by the WiMAX Forum as indicated by the IEEE and reviewed in the ITU-R Working Parties 8F, 8A, 9B, and 9D.
A1.2.27	*Dynamic channel allocation*: characterize the dynamic channel allocation (DCA) schemes which may be supported and characterize their impact on system performance (e.g. in terms of adaptability to varying interference conditions, adaptability to varying traffic conditions, capability to avoid frequency planning, impact on the reuse distance, etc.).	Various permutations of OFDMA subcarriers enable dynamic usage of the spectrum among cells to balance the load and/or average interferences. In a PUSC channel (see A1.2.4), all subcarriers are divided into six groups, some of which are allocated to a particular sector and cell. Although at least one group is assigned to each three cell nearby or 3 sector, remains can be assigned as flexible.

A1.2.28	*Mixed cell architecture*: how well does the RTT accommodate mixed cell architectures (pico, micro and macrocells)? Does the proposal provide pico, micro and macro cell user service in a single licensed spectrum assignment, with handoff as required between them? (terrestrial component only). NOTE 1 – Cell definitions are as follows: – pico – cell hex radius: $r < 100$ m – micro: $100 \text{ m} < r < 1\,000$ m – macro: $r > 1\,000$ m.	The proposed RTT can support flexible frequency reuse operation thus mixed cell architecture is supported well on the same or different frequencies depending on the implementation.
A1.2.29	Describe any battery saver/intermittent reception capability.	
A1.2.29.1	*Ability of the MS to conserve standby battery power*: provide details about how the proposal conserves standby battery power.	[See [1] 6.3.21 Sleep Mode, 6.3.24 Idle Mode] The battery power saving of mobile station is supported by the sleep mode and the idle mode operations. Since the RTT basically provides packet-based transmission, both two modes operate in a slotted mode. In those modes, a mobile station communicates to its serving base station only in a listening interval and saves its power consumption otherwise. The information on listening, sleep and idle intervals are determined by the negotiation between the base station and the mobile station before the mobile station transits to either of two modes. A mobile station maintains the connection to its serving base station even in the sleep mode, while a mobile station in the idle mode returns system resources relevant to the existing connection to a base station. In latter case, the mobile station is managed by the multiple base stations grouped in a paging zone.
A1.2.30	*Signalling transmission scheme*: if the proposed system will use RTTs for signalling transmission different from those for user data transmission, describe the details of the signalling transmission scheme over the radio interface between terminals and base (satellite) stations.	The same RTT is used for both user data and signalling transmission.
A1.2.30.1	Describe the different signalling transfer schemes which may be supported, e.g. in connection with a call, outside a call. Does the RTT support: – new techniques? Characterize. – Signalling enhancements for the delivery of multimedia services? Characterize.	Flexible message-based signalling scheme is used. See system description for details.

A1.2.31	Does the RTT support a bandwidth on demand (BOD) capability? BOD refers specifically to the ability of an end-user to request multi-bearer services. Typically, this is given as the capacity in the form of bits per second of throughput. Multi-bearer services can be implemented by using such technologies as multi-carrier, multi-time slot or multi-codes. If so, characterize these capabilities. NOTE 1 – BOD does not refer to the self-adaptive feature of the radio channel to cope with changes in the transmission quality (see § A1.2.5.1).	[See [1] 6.3.6 Bandwidth Allocation and Request mechanism, 6.3.7.3 DL-MAP, 6.3.7.4 UL-MAP, 8.4.4 Frame Structure] The scheduling service is provided for both downlink and uplink traffic. In order for the scheduler to make an efficient resource allocation and provide the desired QoS and data rate in the uplink, mobile stations must feedback accurate and timely information as to the traffic conditions and QoS requirements. To this end, multiple uplink bandwidth request mechanisms, such as bandwidth request through ranging channel, piggyback request and polling are provided to support uplink bandwidth requests. Frequency and time resource allocation in both downlink and uplink is on per frame basis as as to duly react the traffic and channel conditions. Additionally, the amount of resource in each allocation can range from one slot to the entire frame.
A1.2.32	Does the RTT support channel aggregation capability to achieve higher user bit rates?	No
A1.3	Expected performances.	
A1.3.1	For terrestrial test environment only.	
A1.3.1.1	What is the achievable BER floor level (for voice)? NOTE 1 – The BER floor level is evaluated under the BER measuring conditions defined in Annex 2 using the data rates indicated in § 1 of Annex 2.	Coded BER floor is implementation-dependent but achievable floor is significantly below GoS requirements (10^{-3}) within the specified ranges of tolerable delay spread (20us) and Doppler shifts (250Hz).
A1.3.1.2	What is the achievable BER floor level (for data)? NOTE 1 – The BER floor level is evaluated under the measuring conditions defined in Annex 2 using the data rates indicated in § 1 of Annex 2.	Coded BER floor is implementation-dependent but achievable floor is significantly below GoS requirements (10^{-6}) within the specified ranges of tolerable delay spread (20us) and Doppler shifts (250Hz).
A1.3.1.3	What is the maximum tolerable delay spread (ns) to maintain the voice and data service quality requirements? NOTE 1 – The BER is an error floor level measured with the Doppler shift given in the BER measuring conditions of Annex 2.	The maximum specified range of delay spread (20us in Vehicular B) can be tolerated without an equalizer.
A1.3.1.4	What is the maximum tolerable Doppler shift (Hz) to maintain the voice and data service quality requirements? NOTE 1 – The BER is an error floor level measured with the delay spread given in the BER measuring conditions of Annex 2.	At least 500 Hz, based on the observation that Doppler frequency shows about 570Hz for 250 kmph at 2.5GHz
A1.3.1.5	*Capacity* : the capacity of the radio transmission technology has to be evaluated assuming the deployment models described in Annex 2 and technical parameters from § A1.2.22 through § A1.2.23.2.	

A1.3.1.5.1	What is the voice traffic capacity per cell (not per sector): provide the total traffic that can be supported by a single cell (E/MHz/cell) in a total available assigned non-contiguous bandwidth of 30 MHz (15 MHz forward/15 MHz reverse) for FDD mode or contiguous bandwidth of 30 MHz for TDD mode. Provide capacities for all penetration values defined in the deployment model for the test environment in Annex 2. The procedure to obtain this value is described in Annex 2. The capacity supported by not a standalone cell but a single cell within contiguous service area should be obtained here.	See Annex C for details Voice capacity (ITUv pathloss model, PB3 channel model): - 90 Erlangs/MHz/cell for 133, SIMO,10MHz PUSC
A1.3.1.5.2	What is the information capacity per cell (not per sector): provide the total number of user-channel information bits which can be supported by a single cell (Mbit/s/MHz/cell) in a total available assigned non-contiguous bandwidth of 30 MHz (15 MHz forward/15 MHz reverse) for FDD mode or contiguous bandwidth of 30 MHz for TDD mode. Provide capacities for all penetration values defined in the deployment model for the test environment in Annex 2. The procedure to obtain this value is described in Annex 2. The capacity supported by not a standalone cell but a single cell within contiguous service area should be obtained here.	See reference [4] for details : Data capacity (PUSC, ITUv, 60% PB3, 30% VA30, 10% VA120, DL:UL=28:9) - DL SIMO/MIMO = 3.57/5.52 Mbit/s/MHz/cell respy - UL SIMO/MIMO = 1.59/2.1 Mbit/s/MHz/cell respectively
A1.3.1.6	Does the RTT support sectorization? If yes, provide for each sectorization scheme and the total number of user-channel information bits which can be supported by a single site (Mbit/s/MHz) (and the number of sectors) in a total available assigned non-contiguous bandwidth of 30 MHz (15 MHz forward/15 MHz reverse) in FDD mode or contiguous bandwidth of 30 MHz in TDD mode.	Yes, the RTT supports sectorization. The sectorization and frequency reuse schemes are implementation-dependent and consequently, so are the capacities achieved. The tri-sector scheme is the typical scenario with frequency reuse 1 or reuse 3.
A1.3.1.7	*Coverage efficiency*: the coverage efficiency of the radio transmission technology has to be evaluated assuming the deployment models described in Annex 2.	
A1.3.1.7.1	What is the base site coverage efficiency (km^2/site) for the lowest traffic loading in the voice only deployment model? Lowest traffic loading means the lowest penetration case described in Annex 2.	The coverage is larger than the values in the Link Budget in Annex C
A1.3.1.7.2	What is the base site coverage efficiency (km^2/site) for the lowest traffic loading in the data only deployment model? Lowest traffic loading means the lowest penetration case described in Annex 2.	See Link Budget in Annex C
A1.3.2	For satellite test environment only	
A1.3.2.1	What is the required C/N_0 to achieve objective performance defined in Annex 2?	
A1.3.2.2	What are the Doppler compensation method and residual Doppler shift after compensation?	
A1.3.2.3	*Capacity*: the spectrum efficiency of the radio transmission technology has to be evaluated assuming the deployment models described in Annex 2.	
A1.3.2.3.1	What is the voice information capacity per required RF bandwidth (bit/s/Hz)?	
A1.3.2.3.2	What is the voice plus data information capacity per required RF bandwidth (bit/s/Hz)?	
A1.3.2.4	*Normalized power efficiency*: the power efficiency of the radio transmission technology has to be evaluated assuming the deployment models described in Annex 2.	
A1.3.2.4.1	What is the supported information bit rate per required carrier power-to-noise density ratio for the given channel performance under the given interference conditions for voice?	
A1.3.2.4.2	What is the supported information bit rate per required carrier power-to-noise density ratio for the given channel performance under the given interference conditions for voice plus data?	
A1.3.3	*Maximum user bit rate (for data)*: specify the maximum user bit rate (kbit/s) available in the deployment models described in Annex 2.	The maximum bit rates are well above 20160 kbps. (DL/UL ratio = 2:1, PUSC, 64QAM, 5/6 coding rate)

A1.3.4	What is the maximum range (m) between a user terminal and a BS (prior to hand-off, relay, etc.) under nominal traffic loading and link impairments as defined in Annex 2?	See Link Budget in Annex C. The maximum range depends on the deployment and the QoS of a connection
A1.3.5	Describe the capability for the use of repeaters.	Repeaters can be used. There is nothing in the technology that precludes the use of repeaters.
A1.3.6	*Antenna systems* : fully describe the antenna systems that can be used and/or have to be used; characterize their impacts on systems performance, (terrestrial only); e.g., does the RTT have the capability for the use of: – remote antennas: describe whether and how remote antenna systems can be used to extend coverage to low traffic density areas; – distributed antennas: describe whether and how distributed antenna designs are used, and in which IMT-2000 test environments; – Smart antennas (e.g., switched beam, adaptive, etc.): describe how smart antennas can be used and what is their impact on system performance; – other antenna systems.	The air-interface does not place any restrictions on the types of antenna systems. In particular, there is extensive support for a full range of smart antenna technologies, including beamforming, Transmit/Receive diversity and MIMO. The uses of remote and distributed antennas are not precluded.
A1.3.7	Delay (for voice)	Voice services are provided in the PS-domain with appropriate QoS setting (UGS, rtPS or ErtPS)
A1.3.7.1	What is the radio transmission processing delay due to the overall process of channel coding, bit interleaving, framing, etc., not including source coding? This is given as transmitter delay from the input of the channel coder to the antenna plus the receiver delay from the antenna to the output of the channel decoder. Provide this information for each service being provided. In addition, a detailed description of how this parameter was calculated is required for both the uplink and the downlink.	The minimum delay is roughly 10ms assuming a 5ms TDD frame and the maximum is implementation and traffic load-dependent (scheduling metric, traffic load, buffer sizes, retransmission scheme etc)
A1.3.7.2	What is the total estimated round trip delay (ms) to include both the processing delay, propagation delay (terrestrial only) and vocoder delay? Give the estimated delay associated with each of the key attributes described in Fig. 6 that make up the total delay provided.	Assuming a 20ms vocoder, 5ms frame and ignoring queuing delay (typically <30ms), the RTD delay is approximately 60ms
A1.3.7.3	Does the proposed RTT need echo control?	Yes
A1.3.8	What is the MOS level for the proposed codec for the relevant test environments given in Annex 2? Specify its absolute MOS value and its relative value with respect to the MOS value of ITU-T Recommendation G.711 (64 k PCM) and ITU-T Recommendation G.726 (32 k ADPCM). NOTE 1 – If a special voice coding algorithm is indispensable for the proposed RTT, the proponent should declare detail with its performance of the codec such as MOS level. (See § A1.2.19)	The RTT supports VoIP and is not limited to any particular codecs. Applications/implementations determine the choice of codec.
A1.3.9	Description of the ability to sustain quality under certain extreme conditions.	
A1.3.9.1	*System overload (terrestrial only)* : characterize system behaviour and performance in such conditions for each test services in Annex 2, including potential impact on adjacent cells. Describe the effect on system performance in terms of blocking grade of service for the cases that the load on a particular cell is 125%, 150%, 175%, and 200% of full load. Also describe the effect of blocking on the immediate adjacent cells. Voice service is to be considered here. Full load means a traffic loading which results in 1% call blocking with the BER of 1×10^{-3} maintained.	The RTT provides many features that can be used to ensure optimal loading in the event of system overload. Among these are admission control, handover, rate adaptation, fractional frequency reuse and power control.
A1.3.9.2	*Hardware failures* : characterize system behaviour and performance in such conditions. Provide detailed explanation on any calculation.	This is implementation-dependent. The RTT does not preclude any means to build in redundancy or other reliability features.

A1.3.9.3	*Interference immunity*: characterize system immunity or protection mechanisms against interference. What is the interference detection method? What is the interference avoidance method?	In addition to frequency reuse, and intelligent scheduling/RRM, the RTT's TDD OFDM interface is inherently robust against delay spread, suitable for multi-user detection and supports various smart antenna schemes. Also, the RTT does not preclude any means to cancel interference or to protect against interference
A1.3.10	Characterize the adaptability of the proposed RTT to different and/or time-varying conditions (e.g. propagation, traffic, etc.) that are not considered in the above attributes of § A1.3.	The RTT supports modulation and coding adaptation, HARQ, power control and opportunistic scheduling
A1.4	Technology design constraints	
A1.4.1	*Frequency stability*: provide transmission frequency stability (not oscillator stability) requirements of the carrier (include long term – 1 year – frequency stability requirements (ppm)).	
A1.4.1.1	For BS transmission (terrestrial component only).	*BS frequency tolerance °¬ ± 2ppm of carrier frequency* *BS to BS frequency accuracy °¬ ± 1% of subcarrier spacing*
A1.4.1.2	For MS transmission.	MS to BS frequency synchronization tolerance °¬ 2% of the subcarrier spacing
A1.4.2	*Out-of-band and spurious emissions*: specify the expected levels of base or satellite and mobile transmitter emissions outside the operating channel, as a function of frequency offset.	Base stations and terminals supporting this RTT will comply with local, regional, and international regulations for out of band and spurious emissions, wherever applicable. Similar to other IMT-2000 RTTs, terminals adhering to a single global mask will be used to provide global roaming.

A1.4.3	*Synchronisation requirements*: describe RTT's timing requirements, e.g. – Is BS-to-BS or satellite land earth station (LES)-to-LES synchronisation required? Provide precise information, the type of synchronisation, i.e., synchronisation of carrier frequency, bit clock, spreading code or frame, and their accuracy. – Is BS-to-network synchronisation required? (terrestrial only). – State short-term frequency and timing accuracy of BS (or LES) transmit signal. – State source of external system reference and the accuracy required, if used at BS (or LES) (for example: derived from wireline network, or GPS receiver). – State free run accuracy of MS frequency and timing reference clock. – State base-to-base bit time alignment requirement over a 24 h period (μs).	BS-to-BS synchronisation : Yes. All BSs should be time and frequency synchronized to a common source signal. The common source signal is typically provided by GPS. BS-to-network synchronisation: No. BS-to-network synchronisation is not required. Frequency accuracy : BS frequency tolerance °¬ ± 2ppm of carrier frequency Timing accuracy °¬ 1usec compared to reference timing. Source of external system reference and the accuracy : GPS(the synchronizing reference shall be a 1 pps timing pulse and a 10 MHz frequency reference) Free run accuracy : MS frequency tolerance °¬ maximum 2% of the subcarrier spacing Timing tolerance: 25% of minimum guard interval(± (Tb/32)/4) The BS's timing accuracy is required to be 1•Ï sec compared to reference timing when GPS locked.
A1.4.4	*Timing jitter*: for BS (or LES) and MS give: – the maximum jitter on the transmit signal, – the maximum jitter tolerated on the received signal. Timing jitter is defined as r.m.s. value of the time variance normalized by symbol duration.	**BS** The BS's timing accuracy is required to be 1•Ï sec compared to reference timing. **MS** MS Transmit symbol timing accuracy within ± (Tb/32)/4
A1.4.5	*Frequency synthesizer*: what is the required step size, switched speed and frequency range of the frequency synthesizer of MSs?	Frequency step size : 250 KHz Switched speed : 200 •Ï sec Frequency range : 3.5, 5, 7, 8.75, 10 MHz Start frequencies are various, depending on channel bandwidth and profile
A1.4.6	Does the proposed system require capabilities of fixed networks not generally available today?	No

A1.4.6.1	Describe the special requirements on the fixed networks for the handover procedure. Provide handover procedure to be employed in proposed RTT in detail.	The RTT supports handover and also provides means for expediting handover. Each base station broadcasts the information on the list of neighboring base stations and their channel information such as the operating center frequency, preamble index and synchronization periodically. The channel information in this broadcasting is used for a mobile station to synchronize with the neighboring base station. After a mobile station monitors the signal strength of a neighboring base station and seeks suitable base station(s) for handover, the mobile station or its serving base station can initiate handover by handover request message. But only the mobile station can transmit handover indication message to the its serving base station. After transmitting handover indication message, the mobile station stops monitoring the downlink frame of its serving base station and performs network re-entry to target base station. To reduce the handover latency further, the serving base station provides the target base station with network entry information on a mobile station to be handed over the target base station.
A1.4.7	Fixed network feature transparency	
A1.4.7.1	Which service(s) of the standard set of ISDN bearer services can the proposed RTT pass to users without fixed network modification.	Convergence Sublayer in the proposed RTT supports interface to various fixed networks such as ATM, Ethernet, IP, and VLAN.
A1.4.8	Characterize any radio resource control capabilities that exist for the provision of roaming between a private (e.g., closed user group) and a public IMT-2000 operating environment.	Handover between the different access networks is basically supported. Furthermore, Operator ID in the signalling during the handover enable mobile stations to recognize the operator of access network they are handed over to.
A1.4.9	Describe the estimated fixed signalling overhead (e.g., broadcast control channel, power control messaging). Express this information as a percentage of the spectrum which is used for fixed signalling. Provide detailed explanation on your calculations.	The fixed MAP overhead is typically about 10% in a 10MHz channel with a 5ms frame size.

A1.4.10	Characterize the linear and broadband transmitter requirements for BS and MS (terrestrial only).	**BS** - Tx dynamic Range = 10 dB - Spectral flatness according to the following: °¬ °æ2 dB for spectral lines from °© Nused/4 to 1 and +1 to Nused/4 Within +2/-4 dB for spectral lines from – Nused/2 to Nused/4 and +Nused/4 to Nused/2 - Per sub-carrier flatness °¬ 0.1 dB - Power difference between adjacent subcarriers according to the following: Tx downlink radio frame shall be time-aligned with the 1pps timing pulse within 1 •Ï sec - Tx relative constellation error according to the following: QPSK-1/2 °¬ -15.0 dB QPSK-3/4 °¬ -18.0 dB 16QAM-1/2 °¬ -20.5 dB 16QAM-3/4 °¬ -24.0 dB 64QAM-1/2 (if 64-QAM supported) °¬ -26.0 dB 64QAM-2/3 (if 64-QAM supported) °¬ -28.0 dB 64QAM-3/4 (if 64-QAM supported)°¬ -30.0 dB **MS** - Tx dynamic Range = 45 dB - Tx power level min adjustment step = 1 dB - Tx power level min relative step accuracy = ± 0.5 dB - Spectral flatness according to the following: °¬ °æ2 dB for spectral lines from °©Nused/4 to °©1 and +1 to Nused/4 Within +2/-4 dB for spectral lines from –Nused/2 to –Nused/4 and +Nused/4 to Nused/2 - Power difference between adjacent subcarriers °¬ 0.1 dB - Tx relative constellation error according to the following: QPSK-1/2 °¬ -15.0 dB QPSK-3/4 °¬ -18.0 dB 16QAM-1/2 °¬ -20.5 dB 16QAM-3/4 °¬ -24.0 dB

A1.4.11	Are linear receivers required? Characterize the linearity requirements for the receivers for BS and MS (terrestrial only).	**BS** No. The PAPR of the proposed RTT is around 12dB, and which is not required a stringent linear receiver.
A1.4.12	Specify the required dynamic range of receiver (terrestrial only).	**BS** Max input level on-channel reception tolerance = -45 dBm Max input level on-channel damage tolerance = -10 dBm **MS** Max input level on-channel reception tolerance = -30 dBm Max input level on-channel damage tolerance = 0 dBm Max input level sensitivity (Distributed permutation of subcarriers) -85.1 dBm - QPSK-3/4 -82.8 dBm - 16QAM-1/2 -78.7 dBm - 16QAM-3/4 -77.6 dBm - 64QAM-1/2 -74.5 dBm - 64QAM-2/3 -73.4 dBm - 64QAM-3/4 -71.5 dBm - 64QAM-5/6 Sensitivity numbers are calculated based on assumption of repetition factor 1
A1.4.13	What are the signal processing estimates for both the handportable and the BS? – MOPS (millions of operations per second) value of parts processed by DSP (digital signal processing), – gate counts excluding DSP, – ROM size requirements for DSP and gate counts (kbytes), – RAM size requirements for DSP and gate counts (kbytes). NOTE 1 – At a minimum the evaluation should review the signal processing estimates (MOPS, memory requirements, gate counts) required for demodulation, equalization, channel coding, error correction, diversity processing (including Rake receivers), adaptive antenna array processing, modulation, A-D and D-A converters and multiplexing as well as some IF and baseband filtering. For new technologies, there may be additional or alternative requirements (such as FFTs etc.). NOTE 2 – The signal processing estimates should be declared with the estimated condition such as assumed services, user bit rate and etc.	It is an implementation issue not covered by the description.
A1.4.14	*Dropped calls* : describe how the RTT handles dropped calls. Does the proposed RTT utilize a transparent reconnect procedure – that is, the same as that employed for handoff?	No specific process to handle call dropping recovery is defined. However, mobile station can recover the connection after call dropping by means of the Idle mode re-entry procedure.

A1.4.15	Characterize the frequency planning requirements: – frequency reuse pattern: given the required C/I and the proposed technologies, specify the frequency cell reuse pattern (e.g. 3-cell, 7-cell, etc.) and, for terrestrial systems, the sectorization schemes assumed; – characterize the frequency management between different cell layers; – does the RTT use an interleaved frequency plan? – are there any frequency channels with particular planning requirements? – all other relevant requirements. NOTE 1 – The use of the second adjacent channel instead of the adjacent channel at a neighbouring cluster cell is called "interleaved frequency planning". If a proponent is going to employ an interleaved frequency plan, the proponent should state so in § A1.2.4 and complete § A1.2.15 with the protection ratio for both the adjacent and second adjacent channel.	The RTT supports frequency reuse configuration of 1 and 3. In order for MS to provide BS with a correct DL channel quality information, MS is required to properly measure CINR of preamble with considering the frequency reuse configuration: i.e. For frequency reuse of 3, consider the modulated subcarriers of the preamble only. For frequency reuse of 1, consider both the unmodulated and the modulated subcarriers of the preamble. There are 114 different preamble code sets in the proposed RTT to differentiate the cell ID and sector ID's per each sector. The RTT can use both the interleaved frequency plan and the non-interleaved frequency plan.
A1.4.16	Describe the capability of the proposed RTT to facilitate the evolution of existing radio transmission technologies used in mobile telecommunication systems migrate toward this RTT. Provide detail any impact and constraint on evolution.	
A1.4.17	Are there any special requirements for base site implementation? Are there any features which simplify implementation of base sites? (terrestrial only)	No
A1.5	Information required for terrestrial link budget template Proponents should fulfil the link budget template given in Table 6 and answer the following questions.	see Annex C Link Budget
A1.5.1	What is the BS noise figure (dB)?	4 dB (see Annex C Link Budget)
A1.5.2	What is the MS noise figure (dB)?	7 dB (see Annex C Link Budget)
A1.5.3	What is the BS antenna gain (dBi)?	15 dBi (see Annex C Link Budget)
A1.5.4	What is the MS antenna gain (dBi)?	-1 dBi (see Annex C Link Budget)
A1.5.5	What is the cable, connector and combiner losses (dB)?	0 dB (see Annex C Link Budget)
A1.5.6	What are the number of traffic channels per RF carrier?	Function of required QoS
A1.5.7	What is the RTT operating point (BER/FER) for the required E_b/N_0 in the link budget template?	1% FER
A1.5.8	What is the ratio of intra-sector interference to sum of intra-sector interference and inter-sector interference within a cell (dB)?	Depends on environment and receiver implementation
A1.5.9	What is the ratio of in-cell interference to total interference (dB)?	Negligible at low doppler (<300 Hz) and depends on receiver implementation at high doppler

A1.5.10	What is the occupied bandwidth (99%) (Hz)?	Depends on nominal bandwidth, permutation scheme, and on the sub-channelization. For the case considered in Annex C Link Budget, it is approximately 9.2 MHz on the downlink and 2.4 MHz on the uplink
A1.5.11	What is the information rate (dBHz)?	Depends on service rate with the maximum subject to the channel bandwidth employed. (see Annex C Link Budget)

Annex B. Requirements & Objectives Template

Table 1

Technical Requirements and Objectives Relevant to the Evaluation of Candidate Radio Transmission Technologies

IMT-2000 Item Description	Obj/Req	Source	Meets?*
Voice and data performance requirements			
One-way end to end delay less than 40 ms	Req	G.174, § 7.5	r Yes r No Yes
For mobile videotelephone services, the IMT-2000 terrestrial component should operate so that the maximum overall delay (as defined in ITU-T Rec. F.720) should not exceed 400 ms, with the one way delay of the transmission path not exceeding 150 ms	Req	Suppl. F.720, F.723, G.114	r Yes r No Yes
Speech quality should be maintained during ≤ 3% frame erasures over any 10 second period. The speech quality criterion is a reduction of ≤ 0.5 mean opinion score unit (5 point scale) relative to the error-free condition (G.726 at 32 kb/s)	Req	G.174, § 7.11 & M.1079 § 7.3.1	r Yes r No Yes
DTMF signal reliable transport (for PSTN is typically less than one DTMF errored signal in 10^4)	Req	G.174, § 7.11 & M.1079 § 7.3.1	r Yes r No Yes

* Explanation is requested when the candidate SRTT checks the No box.

Voiceband data support including G3 facsimile	Req	M.1079 § 7.2.2	r Yes r No Yes
Support packet switched data services as well as circuit switched data; requirements for data performance given in ITU-T G.174 *Note: The RTT is purely a Packet switch data technology. Circuit switched data is not supported. But will support seamless interworking with circuit switched systems using media gateways and support for QoS classes.*	Req	M.1034 §§ 10.8, 10.9	r Yes r No Yes and No
Radio interfaces and subsystems, network related performance requirements			
Network interworking with PSTN and ISDN in accordance with Q.1031 and Q.1032	Req	M.687-1 § 5.4	r Yes r No Yes
Meet spectral efficiency and radio channel performance requirements of M.1079	Req	M.1034 § 12.3.3/4	r Yes r No Yes
Provide phased approach with data rates up to 2 Mbit/s in phase 1	Obj	M.687, § 1.1.14	r Yes r No Yes
Maintain bearer channel bit-count integrity (e.g. synchronous data services and many encryption techniques)	Obj	M.1034, § 10.12	r Yes r No Yes
Support for different cell sizes, for example - Mega cell Radius ~ 100 - 500 km Macro cell Radius ≤ 35km, Speed ≤ 500 km/h Micro cell Radius ≤ 1 km, Speed ≤ 100 km/h Pico cell Radius ≤ 50m, Speed ≤10 km/h	Obj	M.1035 § 10.1	r Yes r No Yes
Application of IMT-2000 for fixed services and developing countries			
Circuit noise - idle noise levels in 99% of the time about 100 pWp	Obj	M.819-1, § 10.3	r Yes r No Yes
Error performance - as specified in ITU-R F.697	Obj	M.819-1, § 10.4	r Yes r No Yes
Grade of service better than 1%	Obj	M.819-1, § 10.5	r Yes r No Yes

Table 2. Generic Requirements and Objectives Relevant to the Evaluation of Candidate Radio Transmission Technologies

IMT-2000 Item Description	Obj/Req	Source	Meets?*
Radio interfaces and subsystems, network related performance requirements			
Security comparable to that of PSTN/ISDN	Obj	M.687-1 § 4.4	❐ Yes ❐ No Yes
Support mobility, interactive and distribution services	Req	M.816 § 6	❐ Yes ❐ No Yes
Support UPT and maintain common presentation to users	Obj	M.816 § 4	❐ Yes ❐ No
Voice quality comparable to the fixed network (applies to both mobile and fixed service)	Req	M.819-1 Table 1, M.1079 § 7.1	❐ Yes ❐ No Yes
Support encryption and maintain encryption when roaming and during handover	Req	M.1034 § 11.3	❐ Yes ❐ No Yes
Network access indication similar to PSTN (e.g. dialtone) *Note: These are application specific and not mandated by the RTT. But applications may support this.*	Req	M.1034 § 11.5	❐ Yes ❐ No Yes
Meet safety requirements and legislation	Req	M.1034 § 11.6	❐ Yes ❐ No Yes
Meet appropriate EMC regulations	Req	M.1034 § 11.7	❐ Yes ❐ No Yes
Support multiple public/private/ residential IMT-2000 operators in the same locality	Req	M.1034 § 12.1.2	❐ Yes ❐ No Yes
Support multiple mobile station types	Req	M.1034 § 12.1.4	❐ Yes ❐ No Yes
Support roaming between IMT-2000 operators and between different IMT-2000 radio interfaces/ environments	Req	M.1034 § 12.2.2	❐ Yes ❐ No Yes
Support seamless handover between different IMT-2000 environments such that service quality is maintained and signalling is minimized	Req	M.1034 § 12.2.3	❐ Yes ❐ No Yes

* Explanation is requested when the candidate SRTT checks the No box.

Simultaneously support multiple cell sizes with flexible base location, support use of repeaters and umbrella cells as well as deployment in low capacity areas	Req	M.1034 § 12.2.5	❒ Yes ❒ No Yes
Support multiple operator coexistence in a geographic area	Req	M.1034 § 12.2.5	❒ Yes ❒ No Yes
Support different spectrum and flexible band sharing in different countries including flexible spectrum sharing between different IMT-2000 operators (see M.1036)	Req	M.1034 § 12.2.8	❒ Yes ❒ No Yes
Support mechanisms for minimizing power and interference between mobile and base stations	Req	M.1034 § 12.2.8.3	❒ Yes ❒ No Yes
Support various cell types dependent on environment (M.1035 § 10.1)	Req	M.1034 § 12.2.9	❒ Yes ❒ No Yes
High resistance to multipath effects	Req	M.1034 § 12.3.1	❒ Yes ❒ No Yes
Support appropriate vehicle speeds (as per § 7) NOTE: applicable to both terrestrial and satellite proposals	Req	M.1034 § 12.3.2	❒ Yes ❒ No Yes
Support possibility of equipment from different vendors	Req	M.1034 § 12.1.3	❒ Yes ❒ No Yes
Offer operational reliability as least as good as 2nd generation mobile systems	Req	M.1034 § 12.3.5	❒ Yes ❒ No Yes
Ability to use terminal to access services in more than one environment, desirable to access services from one terminal in all environments	Obj	M.1035 § 7.1	❒ Yes ❒ No Yes
End-to-end quality during handover comparable to fixed services	Obj		❒ Yes ❒ No Yes
Support multiple operator networks in a geographic area without requiring time synchronization	Obj		❒ Yes ❒ No Yes
Layer 3 contains functions such as call control, mobility management and radio resource management some of which are radio dependent. It is desirable to maintain layer 3 radio transmission independent as far as possible	Obj	M.1035 § 8	❒ Yes ❒ No Yes

Desirable that transmission quality requirements from the upper layer to physical layers be common for all services	Obj	M.1035 § 8.1	❒ Yes ❒ No Yes
The link access control layer should as far as possible not contain radio transmission dependent functions	Obj	M.1035 § 8.3	❒ Yes ❒ No Yes
Traffic channels should offer a functionally equivalent capability to the ISDN B-channels	Obj	M.1035 § 9.3.2	❒ Yes ❒ No Yes
Continually measure the radio link quality on forward and reverse channels	Obj	M.1035 § 11.1	❒ Yes ❒ No Yes
Facilitate the implementation and use of terminal battery saving techniques	Obj	M.1035 § 12.5	❒ Yes ❒ No Yes
Accommodate various types of traffic and traffic mixes	Obj	M.1036 § 1.10	❒ Yes ❒ No Yes
Application of IMT-2000 for fixed services and developing countries			
Repeaters for covering long distances between terminals and base stations, small rural exchanges with wireless trunks etc.	Req	M.819-1 Table 1	❒ Yes ❒ No Yes
Withstand rugged outdoor environment with wide temperature and humidity variations	Req	M.819-1 Table 1	❒ Yes ❒ No Yes
Provision of service to fixed users in either rural or urban areas	Obj	M.819-1 § 4.1	❒ Yes ❒ No Yes
Coverage for large cells (terrestrial)	Obj	M.819-1 § 7.2	❒ Yes ❒ No Yes
Support for higher encoding bit rates for remote areas	Obj	M.819-1 § 10.1	❒ Yes ❒ No Yes
Additional satellite- component specific requirements and objectives			
Links between the terrestrial and satellite control elements for handover and exchange of other information	Req	M.818-1 § 3.0	❒ Yes ❒ No NA

Take account for constraints for sharing frequency bands with other services (WARC-92)	Obj	M.818-1 § 4.0	❒ Yes ❒ No NA
Compatible multiple access schemes for terrestrial and satellite components	Obj	M.818-1 § 6.0	❒ Yes ❒ No NA
Service should be comparable quality to terrestrial component as far as possible	Obj	M.818-1 § 10.0	❒ Yes ❒ No NA
Use of satellites to serve large cells for fixed users	Obj	M.819-1 § 7.1	❒ Yes ❒ No NA
Key features (e.g. coverage, optimization, number of systems)	Obj	M.1167 § 6.1	❒ Yes ❒ No NA
Radio interface general considerations	Req	M.1167 § 8.1.1	❒ Yes ❒ No NA
Doppler effects	Req	M.1167 § 8.1.2	❒ Yes ❒ No NA

Annex C. Capacity and Coverage

Voice Capacity

The VoIP capacity results were generated from OPNET simulations based on a 19-cell scenario and detailed modeling of the 802.16e MAC protocols, overhead and latencies. The simulation statistics are collected on the center sector whose loading is controlled by varying the number of users.

Methodology

1. Configuration with N-users in a sector (N is varied).
2. Pick a Nominal SINR at random for each MSS, reflecting the path loss, shadowing and interference
3. Propagate the UL and DL channels for fast fading using ITU Ped-B
4. UGS QoS for the voice service flow
    a. Allocate 32B of grant on an average every 4 frames (20 msec) on the UL
    b. Schedule the voice flows on the DL at an average of every 20msec.
    c. Max queueing latency bound on the transmit queue = 50msec
        i. Packets waiting in the queue, beyond this bound, are dropped.
5. Xrtps QoS modeling for voice activity detection with 40% activity factor
6. Link Adaptation is done via MAC Management messages.
7. For each mobile during the simulation, FER, throughput and latency statistics are collected.
    a. FER accounts for frame drops due to both channel errors and transmit queue latency > 50msec.

Simulation Assumptions

- **Results generated from OPNET MAC-E2E simulator**
- **Pedestrian channel modeling– *frequency selective fast fading***
- **3 sector cell**
- SIMO-EGC (2 Rx antennas, 1 Tx antenna)
- DL:UL 3:2
- 5MHz/10MHz TDD channel
- **1x3x3 PUSC reuse scheme**
- **Target PER = 3%**
- No sub MAPs
- **MLWDF scheduler – channel aware and resource fair**
    - Channel aware and resource fair scheduler.
    - Metric for user *i* at time *t*:
        - $M(i, t) = c(i) * (\mu(i, t)/A(i, t)) * W(i, t)$
        - *Where*
            - $\mu(i, t)$ = *instantaneous rate of the user*
            - $A(i, t) = (\alpha(i)*\mu(i, t) + (1-\alpha(i))*A(i, t-1))$ = *mean rate of user*
            - $W(i, t)$ = *queue length at time t*
            - $c(i)$ = *a constant for user i, indicating his relative priority*
    - Advantage of MLWDF over PF:
        - Adding the queue length to the metric, forces a user with to get served even if the channel is poor for a long time. This increases the overall fairness.
    - Optimized and Proprietary rectangularization algorithms
- **VoIP packet assumptions**
    - RTP/UDP/IP encapsulation
    - G.729 codec: 20B of voice payload every 20msec
    - PHS
        - Part of the 802.16 spec
        - All the RTP/UDP/IP headers can be suppressed except for RTP sequence numbers (2B)
            - CID of the MAC header can be made to uniquely map to the header parameters
        - The resulting RTP/UDP/IP header is 2B

- **Other System parameters**

CellSize	**1.5 km**
PathLoss Model	**ITU Vehicular**
CarrierFrequency	**2.3 GHz**
BSHeight	**30m**
BSTxPower	**36dBm**
BSAntennaGain	**16dB**
3dB antenna beamwidth	70 degrees
SSHeight	**Vehicular(1.5m)**
SSTxPower	**23dBm**
SSAntennaGain	**0dB**

Fig 1 traces the latencies incurred, assuming no queuing delays, from the generation of a voice frame in a MS to its reception at the BS.

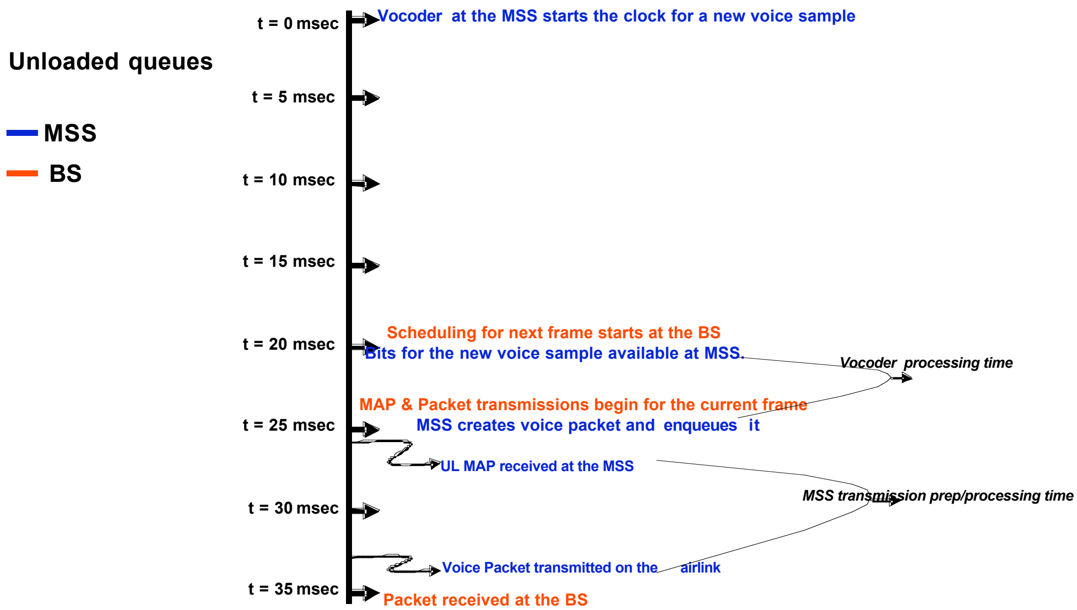

Figure C1. VoIP UL Packet processing timeline

Results

The VoIP capacity is defined here as the Erlangs supported per 3-sector cell at a FER of <3% and a 1-way delay of <100ms (ITU-R M.1079-1). The capacity was FER-limited, with the 1-way latencies (queuing and transmission latencies) at <85ms and typically 55-65ms for the uplink, well under the 100ms bound. The downlink has a lower latency bound because of the absence of scheduling grant latencies.

The voice capacity for the above assumptions was determined to be 90 Erlangs/MHz/cell.

Handover Performance

The 802.16e specifications support 3 types of handover (HO), Hard Handover (HHO), Soft Handover (SHO) and Fast Base Station Selection (FBSS). Handovers may be initiated by either the mobile or the base station based on signal quality, loading or service criteria. The handover results are based on analysis of the protocol specifications. The handover performance metrics used here are the minimum time required to execute handover and the interruption time during the handover process during which time data transfer is stopped. Therefore, HHO is the worst-case of the 3 types of HOs and is the case for which performance is evaluated.

The handover performance requirements are dictated primarily by the QoS requirements. Voice service requires minimum interruption times and delays but can tolerate some loss of data. On the other hand, data services (non-realtime / Best-Effort Data) are more tolerant of interruption and delays but require minimum or no loss of data.

Assumptions and Procedure

The following assumptions were used in deriving the HO performance:

- **5ms frame**
- **MS-assisted-NW-controlled for**
  - **_Neighbor BS prioritizing**
  - **_Inter-frequency scan profiles & reporting criteria**
- **All measurements based on frame preamble**

The Handover procedure, is as follows:

I. Preparation
  1. Handover request from MS to BS (if MS-initiated)
  2. Source base station selects target base station after backbone pre-notification procedure (collocated case optimizations)
  3. BS-MS exchange HO decision

II. HO (Collocated & intra-frequency case optimizations)
  1. HO Ranging in target base station
  2. NW Re-entry

HO Performance Results

The Handover performance metrics are defined as follows:

- **Preparation time - duration between the HO decision and the HO trigger**
- **Interruption time - duration between stopping serving BS TX/RX and starting target BS TX/RX**

The results are summarized in Table C1 for the scenario described below. Table C2 shows the calculation for scenarios 1 and 2.

Scenario 1. MS initiated HHO, intra-FA, non-collocated, full optimized NW re-entry (only RNG-REQ/RSP)
This is a scenario, where NW conditions may be comparable to FBSS (frequency reuse factor 1, TEK sharing between BS's, no ranging), and where HHO break-off duration is very short.. HO is between two separate BS's with same FA's

Scenario 2. MS initiated HHO, intra-FA, collocated, full optimized NW re-entry
Similar to preceding scenario, except HO is between *two PUSC sectors within same BS*. HO latency is same as in the non-collocated case.

Scenario 3. MS initiated HHO, inter-FA, non-collocated, full optimized NW re-entry
Similar to preceding scenario, except HO is between BS's with *different* FA's

Scenario 4.MS initiated HHO, optimized NW re-entry with TEKs update
Similar to preceding scenario, except NW re-entry procedure includes *TEKs update*

Scenario 5. MS initiated HHO due to link loss (drop situation)

- Link loss is MAC synchronization loss:
- 600ms max since last good received DL-MAP or 5*DCD interval max value (10s max) since last good received DCD (the earliest of the two). MS may decide earlier than that to HO.
- MS will HO to 1st HO candidate (from cell selection list, best received NBR BS), and perform CDMA ranging using HO codes.
- TBS identifies HO code in ranging region and may allocate ranging regions more frequently (e.g. every frame).
- Upon receiving RNG-REQ, TBS will request MS context from SBS, and perform optimized NW re-entry (like any coordinated HO).

Scenario 6. BS initiated HHO due to scan result

- All HO decision making is done by Serving BS
- MS reports scan results of neighboring BSs
- BS decides to HO according to scan report

Table C1. Summary of HO latency analysis for optimized HHO

Scenario	Preparation time [ms]	Interruption time [ms]

1. MS initiated HHO, intra-FA, non-collocated, full optimized NW re-entry (only RNG-REQ/RSP)	**70-85**	**45**
2. MS initiated HHO, intra-FA, *collocated*, fully optimized NW re-entry	**70-85**	**45**
3. MS initiated HHO, *inter*-FA, non-collocated, fully optimized NW re-entry.	**70-85**	**50**
4. MS initiated HHO, optimized NW re-entry *with TEKs update*	**60-75**	**80-85**
5. MS initiated HHO due to link loss (drop situation)	**0**	**140-165**
6. BS initiated HHO	**35-50**	**50**

Table C2. HO Duration Analysis for Scenarios 1 and 2

	Action	Duration [frames]	Remarks
1.	MS decides to initiate HO to Target BS	0	• MS has a valid (updated) MOB_NBR_ADV list • MS is able to estimate ranging (Tx PHY) parameters
2.	MS sends MOB_MSHO_REQ to SBS (with preferred TBS)	4-7	If currently allocated user BW is insufficient to accommodate this message, a BW request must be issued first (7 frames, CDMA BW request to UL allocation)
3.	SBS processes MOB_MSHO_REQ and responds with MOB_BSHO_RSP (with recommended TBS)	6	• Assumes that recommended BS list includes MS' preferred TBS (no "reject" expect from MS) • At this time, the SBS informs TBS about MS' intent to HO • SBS sends pre-notification request and receives response within 2 frames • Unsolicited UL allocation by TBS (for MOB_HO-IND).
4.	MS processes MOB_BSHO_RSP and transmits MOB_HO_IND to SBS (with HO type and preferred TBS)	4	• If TBS is different from the TBS in the HO_REQ message, the SBS informs the final TBS about MS' intent to HO to it • From this moment and on, the SBS will retain resources of MS with timeout
5.	MS switches to TBS, acquires DL signal and implements channel estimation result	2	• 1st frame at TBS is used for channel estimation. • In 2nd frame at TBS, MS is ready to read MAPs
6.	TBS allocates FAST_RANGING_IE (at estimated HO time)	0	• TBS may allocate FAST_RANGING_IE in two consecutive frames (for robustness).
7.	MS sends RNG-REQ to TBS (with OMAC)	1	• UL-MAP relevance is next frame • RNG-REQ is CMAC signed.

8.	TBS processes RNG-REQ message and responds with RNG_RSP (with CID update and HO optimization flag)	3	• RNG_RSP message includes SBC_RSP and REG_RSP TLV's (for CID update) • RNG_RSP message includes security related TLV's. • Unsolicited UL allocation by TBS (for MOB_HO-IND).
9.	MS processes RNG-RSP (CID update)	3	• Assumes key sharing between sectors within same BS
10.	TBS starts normal operations with MS	0	
HO preparation period		**14-17**	**70-85 ms (@frame=5ms)**
HO break-off duration		**9**	**45 ms (@frame=5ms)**

Data Capacity

System Parameters

Since Mobile WiMAX is based on scalable OFDMA, it can be flexibly configured to operate on different bandwidths by adjusting system parameters. We consider a Mobile WiMAX system with the following characteristics as a case study for a quantitative evaluation of Mobile WiMAX system performance. In the following tables, Table C3 provides the system parameters, Table C4 summarizes the OFDMA parameters, and Table C5 shows the propagation model used for the performance evaluation.

Table C3. Mobile WiMAX System Parameters

Parameters	Value
Number of 3-Sector Cells	19
Operating Frequency	2500 MHz
Duplex	TDD
Channel Bandwidth	10 MHz
BS-to-BS Distance	2.8 kilometers
Minimum Mobile-to-BS Distance	36 meters
Antenna Pattern	70° (-3 dB) with 20 dB front-to-back ratio
BS Height	32 meters
Mobile Terminal Height	1.5 meters
BS Antenna Gain	15 dBi
MS Antenna Gain	-1 dBi
BS Maximum Power Amplifier Power	43 dBm
Mobile Terminal Maximum PA Power	23 dBm
# of BS Tx/Rx Antenna	Tx: 2 or 4; Rx: 2 or 4
# of MS Tx/Rx Antenna	Tx: 1; Rx: 2
BS Noise Figure	4 dB
MS Noise Figure	7 dB

Table C4. OFDMA Parameters

Parameters		Values
System Channel Bandwidth (MHz)		10
Sampling Frequency (F_p in MHz)		11.2
FFT Size (N_{FFT})		1024
Sub-Carrier Frequency Spacing		10.94 kHz
Useful Symbol Time ($T_b = 1/f$)		91.4 microseconds
Guard Time ($T_g = T_b/8$)		11.4 microseconds
OFDMA Symbol Duration ($T_s = T_b + T_g$)		102.9 microseconds
Frame duration		5 milliseconds
Number of OFDMA Symbols		48
DL PUSC	Null Sub-carriers	184
	Pilot Sub-carriers	120

Parameters		Values
	Data Sub-carriers	720
	Sub-channels	30
UL PUSC	Null Sub-carriers	184
	Pilot Sub-carriers	280
	Data Sub-carriers	560
	Sub-channels	35

Table C5. Propagation Model

Parameters	Value
Propagation Model	COST 231 Suburban
Log-Normal Shadowing SD (_s)	8 dB
BS Shadowing Correlation	0.5
Penetration Loss	10 dB

System Performance

Simulations based on the methodology in [6],[7] have been performed to assess the performance of Mobile WiMAX. The system parameters for the Mobile WiMAX system are described in Tables C3-C5. The performance simulation assumes heterogeneous users with a mix of mobile users as described in Tables C6 and C7.

Table C6. Multi-Path Channel Models for Performance Simulation

Channel Model	Path 1 (dB)	Path 2 (dB)	Path 3 (dB)	Path 4 (dB)	Path 5 (dB)	Path 6 (dB)	Rake Fingers
ITU Ped. B Ch-103	-3.92	-4.82	-8.82	-11.92	-11.72	-27.82	1,2,3,4,5,6
ITU Veh. A Ch-104	-3.14	-4.14	-12.14	-13.14	-18.14	-23.14	1,2,3,4,5,6

Table C7. Mixed User Channel Model for Performance Simulation

Channel Model	Number of Paths	Speed	Fading	Assignment Probability
ITU Ped. B Ch-103	6	3 km/hr	Jakes	0.60
ITU Veh. A Ch-104	6	30 km/hr	Jakes	0.30
	6	120 km/hr	Jakes	0.10

There are 10 users per sector. The traffic is assumed to be full buffer FTP traffic. Proportional Fair scheduler is assumed. Each base station is configured with three (3) sectors with a cell and sector frequency reuse factor equal to one. Ideal channel estimation and realistic link adaptation is also assumed. The carrier frequency for the Mobile WiMAX simulation is 2.5 GHz The frame overhead to account for Preamble, MAP OH, and UL Control Channel is 7 OFDMA symbols in the DL and 3 in the UL. 1 symbol is allocated for TTG for a total of 11 overhead symbols and 37 data symbols for both DL and UL. Further configuration and assumption details are listed in Table C8.

Table C8. Mobile WiMAX Configuration Assumptions

Parameters	Value
Cell Configuration	3 Sectors/Cell
Frequency Reuse	1
Users/Sector	10
Traffic Type	Full Buffer
Channel Estimation	Ideal
PHY Abstraction	EESM [8]
Scheduler	Proprietary Proportional Fair
Link Adaptation	Realistic with delay feedback
Antenna Configuration	1x2, 2x2

Parameters		Value
MIMO Support	DL	Alamouti STC, VSM
	UL	Collaborative SM
MIMO Switch		Adaptive STC/VSM switch
HARQ		CC, 3 Retransmissions
Coding		CTC
Frame Overhead		11 OFDM Symbols (7 DL, 3 UL, 1 TTG)
Data Symbols per Frame		37
DL/UL Partition	A	28:9
	B	22:15

The performance is summarized in Table 15 for a TDD implementation with a 10 MHz channel bandwidth, SIMO and MIMO antenna configurations and DL/UL ratios of 28:9 and 22:15 respectively. The results show that the Mobile WiMAX system has high spectral efficiency. With two receive antennas, the DL sector spectral efficiency is about 1.2 bits/sec/Hz and UL sector spectral efficiency is 0.55 bits/sec/Hz. With 2x2 MIMO, the spectral efficiency is further improved by 55% in the DL and 40% in the UL. The high spectral efficiency combined with wide channel bandwidth provides very high sector throughput for the Mobile WiMAX system. With 2x2 MIMO and a DL/UL ratio of 3:1, the DL sector throughput is 13.60 Mbps and the UL sector throughput is 1.83 Mbps; with a DL/UL ratio of 3:2, the sector throughput is 10.63 Mbps and 2.74 Mbps respectively for DL and UL. The high sector data throughput is essential to enable broadband data services including video and VoIP.

It should be noted that 11 symbols of overhead is a conservative estimate for overhead. For most data applications, the traffic is bursty and WiMAX can operate more efficiently with less overhead. Additionally, the sub-channel considered for this case is PUSC diversity sub-channelizaiton and frequency selective scheduling gain is not taken into account in the simulation. With frequency selective AMC sub-channelization, the spectral efficiency can be further increased by 15 to 25% [9]. Therefore, with an optimized Mobile WiMAX system, the spectral efficiency and throughput can be further improved by 20 to 30% compared to the results shown in Table C9. The spectral efficiency improvement for this case is illustrated in Figure C2 for the 2x2 MIMO antenna configuration.

Table C9. Mobile WiMAX System Performance

Cases		DL: 28 data symbols UL: 9 data symbols		DL: 22 data symbols UL: 15 data symbols	
Antenna	Link	Sector Throughput	Spectral Efficiency	Sector Throughput	Spectral Efficiency
SIMO	DL	8.8 Mbps	1.19 bps/Hz	6.6 Mbps	1.07 bps/Hz
	UL	1.38 Mbps	0.53 bps/Hz	2.20 Mbps	0.57 bps/Hz
MIMO	DL	13.60 Mbps	1.84 bps/Hz	10.63 Mbps	1.73 bps/Hz
	UL	1.83 Mbps	0.70 bps/Hz	3.05 Mbps	0.79 bps/Hz

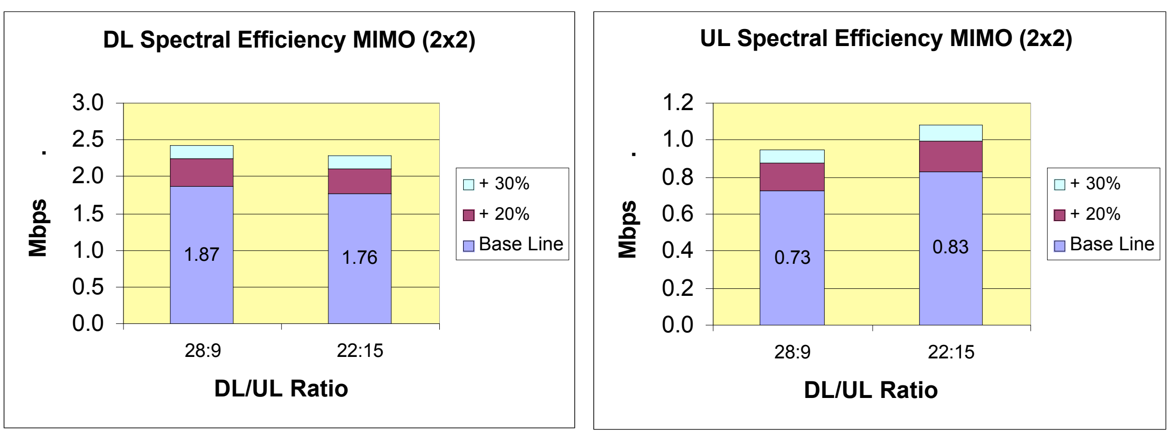

Figure C2. Spectral Efficiency Improvement with Optimized WiMAX

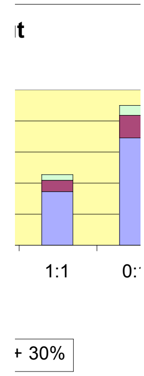

Figure C3. Throughput with Varied DL/UL Ratios and Optimized WiMAX

Another advantage of the Mobile WiMAX system is its ability to dynamically reconfigure the DL/UL ratio to adapt to the network traffic profile so as to maximize spectrum utilization. This is illustrated in Figure 16 where the cross-hatched bars represent the base line values shown in Table 15. It shows that the maximum DL sector throughput can be greater than 20 Mbps and maximum UL sector throughput can be greater than 8 Mbps. With a typical DL/UL ratio range between 3:1 and 1:1, the DL sector throughput can vary between 10 Mbps and 17 Mbps; the UL sector throughput can vary between 2 Mbps and 4 Mbps.

The results here are based on the Mobile WiMAX basic MIMO (2x2) configuration, further performance improvements can be realized with additional advanced Mobile WiMAX features such as AAS.

Link Budget

The 802.16e link budget below is based on the system parameters and channel propagation model in Table C3-C5, for a 2x2 (2-transmit and 2-receive antennas) antenna configuration in the downlink and a 1x2 (1-transmit and 2-receive antennas) antenna configuration in the uplink. The downlink parameters include Cyclic Shift Transmit Diversity and pilot boosting. The value of 5.56 dB used for the Shadow Fade margin in the table assures a 75% coverage probability at the cell edge and 90% coverage probability over the entire area. Note that the maximum allowable path loss, 128.2 dB, corresponds to a DL cell-edge data rate of 5.76 Mbps and an UL cell-edge data rate of 115 kbps. Higher data rate at the cell edge and higher carrier frequency results in smaller cell size. Alternatively, better link budget and larger cell size can be achieved at lower cell edge data rates, as shown in the link budget.

Table C10. Mobile WiMAX Link Budget

Test environment	Outdoor to Indoor			
Test service	UDD(PUSC permutation)			
	Downlink		Uplink	
Bit rate	2.88 Mbps	5.76Mbps	38 kbps	115 kbps
Average TX power per traffic ch. dBm	40	40	23	23
Maximum TX power per traffic ch. dBm	40	40	23	23
Maximum total TX power dBm	40	40	23	23
TX antenna gain dBi	15	15	-1	-1
Cyclic Combining Gain dB	3	3	0	0
Pilot Boosting Gain dB	-0.7	-0.7	0	0
TX EIRP per traffic channel dBm	57.3	57.3	22	22
Total TX EIRP dBm	57.3	57.3	22	22
RX antenna gain dBi	-1	-1	15	15
Receiver noise figure dB	7	7	4	4
Thermal noise density dBm/Hz	-174	-174	-174	-174
RX interference density dBm/Hz	-176.3	-176.3	-174	-174
Total effect. noise + interf. density dBm/Hz -169	-165	-165	-171	-171
Information rate dBHz	64.6	67.6	45.8	50.6
Required Eb/(No+Io) dB	10.5	13	12.6	12.6
RX sensitivity dBm	-89.9	-84.4	-108.5	-109.8
Explicit diversity gain dB	3	3	3	3
Other gain dB (Building penetration)	10	10	10	10
Log-normal fade margin dB	5.56	5.56	5.56	5.56
Maximum path loss dB	133.7	128.2	133	128.2
Maximum range m	436.2	318.9	420.4	319.4
Coverage efficiency km^2/cell	0.6	0.32	0.56	0.32

Annex D Self-Evaluation

Index	Criteria and attributes	Q or q	Gn	Related attributes in Annex 1
A3.1	Spectrum efficiency The following entries are considered in the evaluation of spectrum efficiency:			
A3.1.1	For terrestrial environment			
A3.1.1.1	Voice traffic capacity (E/MHz/cell) in a total available assigned non-contiguous bandwidth of 30 MHz (15 MHz forward/15 MHz reverse) for FDD mode or contiguous bandwidth of 30 MHz for TDD mode. This metric must be used for a common generic continuous voice bearer with characteristics 8 kbit/s data rate and an average BER 1×10^{-3} as well as any other voice bearer included in the proposal which meets the quality requirements (assuming 50% voice activity detection (VAD) if it is used). For comparison purposes, all measures should assume the use of the deployment models in Annex 2, including a 1% call blocking. The descriptions should be consistent with the descriptions under criterion § 6.1.7 – Coverage/power efficiency. Any other assumptions and the background for the calculation should be provided, including details of any optional speech codecs being considered.	Q and q	G1	A1.3.1.5.1

A3.1.1.2	Information capacity (Mbit/s/MHz/cell) in a total available assigned non-contiguous bandwidth of 30 MHz (15 MHz forward/15 MHz reverse) for FDD mode or contiguous bandwidth of 30 MHz for TDD mode. The information capacity is to be calculated for each test service or traffic mix for the appropriate test environments. This is the only measure that would be used in the case of multimedia, or for classes of services using multiple speech coding bit rates. Information capacity is the instantaneous aggregate user bit rate of all active users over all channels within the system on a per cell basis. If the user traffic (voice and/or data) is asymmetric and the system can take advantage of this characteristic to increase capacity, it should be described qualitatively for the purposes of evaluation.	Q and q	G1	A1.3.1.5.2
A3.1.2	For satellite environment These values (§ A3.1.2.1 and A3.1.2.2) assume the use of the simulation conditions in Annex 2. The first definition is valuable for comparing systems with identical user channel rates. The second definition is valuable for comparing systems with different voice and data channel rates.			
A3.1.2.1	Voice information capacity per required RF bandwidth (bit/s/Hz)	Q	G1	A1.3.2.3.1
A3.1.2.2	Voice plus data information capacity per required RF bandwidth (bit/s/Hz)	Q	G1	A1.3.2.3.2
A3.2	Technology complexity – Effect on cost of installation and operation The considerations under criterion § 6.1.2 – Technology complexity apply only to the infrastructure, including BSs (the handportable performance is considered elsewhere).			
A3.2.1	Need for echo control The need for echo control is affected by the round trip delay, which is calculated as shown in Fig. 6. Referring to Fig. 6, consider the round trip delay with the vocoder (D1, ms) and also without that contributed by the vocoder (D2, ms). NOTE 1 – The delay of the codec should be that specified by ITU-T for the common generic voice bearer and if there are any proposals for optional codecs include the information about those also.	Q	G4	A1.3.7.2 A1.3.7.3
A3.2.2	Transmitter power and system linearity requirements NOTE 1 – Satellite e.i.r.p. is not suitable for evaluation and comparison of RTTs because it depends very much on satellite orbit. The RTT attributes in this section impact system cost and complexity, with the resultant desirable effects of improving overall performance in other evaluation criteria. They are as follows.			
A3.2.2.1	Peak transmitter/carrier (P_b) power (not applicable to satellite) Peak transmitter power for the BS should be considered because lower peak power contributes to lower cost. Note that P_b may vary with test environment application. This is the same peak transmitter power assumed in Annex 2, link budget template (Table 6).	Q	G1	A1.2.16.2.1
A3.2.2.2	Broadband power amplifier (PA) (not applicable to satellite) Is a broadband power amplifier used or required? If so, what are the peak and average transmitted power requirements into the antenna as measured in watts.	Q	G1	A1.4.10 A1.2.16.2.1 A1.2.16.2.2 A1.5.5 A1.2.5
A3.2.2.3	Linear base transmitter and broadband amplifier requirements (not applicable to satellite)			

A3.2.2.3.1	Adjacent channel splatter/emission and intermodulation affect system capacity and performance. Describe these requirements and the linearity and filtering of the base transmitter and broadband PA required to achieve them.	q	G3	A1.4.2 A1.4.10
A3.2.2.3.2	Also state the base transmitter and broadband PA (if one is used) peak to average transmitter output power, as a higher ratio requires greater linearity, heat dissipation and cost.	Q and q	G2	A1.4.10 A1.2.16.2.1 A1.2.16.2.2
A3.2.2.4	Receiver linearity requirements (not applicable to satellite) Is BS receiver linearity required? If so, state the receiver dynamic range required and the impact of signal input variation exceeding this range, e.g., loss of sensitivity and blocking.	q	G4	A1.4.11 A1.4.12
A3.2.3	Power control characteristics (not applicable to satellite) Does the proposed RTT utilize transmitter power control? If so, is it used in both forward and reverse links? State the power control range, step size (dB) and required accuracy, number of possible step sizes and number of power controls per second, which are concerned with BS technology complexity.	Q and q	G4	A1.2.22 A1.2.22.1 A1.2.22.2 A1.2.22.3 A1.2.22.4 A1.2.22.5
A3.2.4	Transmitter/receiver isolation requirement (not applicable to satellite) If FDD is used, specify the noted requirement and how it is achieved.	q	G3	A1.2.2 A1.2.2.2 A1.2.2.1
A3.2.5	Digital signal processing requirements			
A3.2.5.1	Digital signal processing can be a significant proportion of the hardware for some radio interface proposals. It can contribute to the cost, size, weight and power consumption of the BS and influence secondary factors such as heat management and reliability. Any digital circuitry associated with the network interfaces should not be included. However any special requirements for interfacing with these functions should be included. This section of the evaluation should analyse the detailed description of the digital signal processing requirements, including performance characteristics, architecture and algorithms, in order to estimate the impact on complexity of the BSs. At a minimum the evaluation should review the signal processing estimates (MOPS, memory requirements, gate counts) required for demodulation, equalization, channel coding, error correction, diversity processing (including Rake receivers), adaptive antenna array processing, modulation, A-D and D-A converters and multiplexing as well as some IF and baseband filtering. For new technologies, there may be additional or alternative requirements (such as FFTs). Although specific implementations are likely to vary, good sample descriptions should allow the relative cost, complexity and power consumption to be compared for the candidate RTTs, as well as the size and the weight of the circuitry. The descriptions should allow the evaluators to verify the signal processing requirement metrics, such as MOPS, memory and gate count, provided by the RTT proponent.	Q and q	G2	A1.4.13

A3.2.5.2	What is the channel coding/error handling for both the forward and reverse links? Provide details and ensure that implementation specifics are described and their impact considered in DSP requirements described in § A3.2.5.1.	q	G4	A1.2.12 A1.4.13
A3.2.6	Antenna systems The implementation of specialized antenna systems while potentially increasing the complexity and cost of the overall system can improve spectrum efficiency (e.g. smart antennas), quality (e.g. diversity), and reduce system deployment costs (e.g. remote antennas, leaky feeder antennas). NOTE 1 – For the satellite component, diversity indicates the number of satellites involved; the other antenna attributes do not apply.			
A3.2.6.1	*Diversity*: describe the diversity schemes applied (including micro and macro diversity schemes). Include in this description the degree of improvement expected, and the number of additional antennas and receivers required to implement the proposed diversity design beyond and omni-directional antenna.	Q	G2	A1.2.23 A1.2.23.1 A1.2.23.2
A3.2.6.2	*Remote antennas*: describe whether and how remote antenna systems can be used to extend coverage to low traffic density areas.	q	G2	A1.3.6
A3.2.6.3	*Distributed antennas*: describe whether and how distributed antenna designs are used.	q	G3	A1.3.6
A3.2.6.4	*Unique antenna*: describe additional antenna systems which are either required or optional for the proposed system, e.g., beam shaping, leaky feeder. Include in the description the advantage or application of the antenna system.	q	G4	A1.3.6
A3.2.7	BS frequency synchronization/time alignment requirements Does the proposed RTT require base transmitter and/or receiver station synchronization or base-to-base bit time alignment? If so, specify the long term (1 year) frequency stability requirements, and also the required bit-to-bit time alignment. Describe the means of achieving this.	Q and q	G3	A1.4.1 A1.4.3
A3.2.8	The number of users per RF carrier/frequency channel that the proposed RTT can support affects overall cost – especially as bearer traffic requirements increase or geographic traffic density varies widely with time. Specify the maximum number of user channels that can be supported while still meeting ITU-T Recommendation G.726 performance requirements for voice traffic.	Q	G1	A1.2.17
A3.2.9	Base site implementation/installation requirements (not applicable to satellite) BS size, mounting, antenna type and height can vary greatly as a function of cell size, RTT design and application environment. Discuss its positive or negative impact on system complexity and cost.	q	G1	A1.4.17
A3.2.10	Handover complexity Consistent with handover quality objectives defined in criterion § 6.1.3, describe how user handover is implemented for both voice and data services and its overall impact on infrastructure cost and complexity.	Q and q	G1	A1.2.24 A1.4.6.1

A3.3	Quality			
A3.3.1	Transparent reconnect procedure for dropped calls Dropped calls can result from shadowing and rapid signal loss. Air interfaces utilizing a transparent reconnect procedure – that is, the same as that employed for hand-off – mitigate against dropped calls whereas RTTs requiring a reconnect procedure significantly different from that used for hand-off do not.	q	G2	A1.4.14
A3.3.2	Round trip delay, D1 (with vocoder (ms)) and D2 (without vocoder (ms)) (See Fig. 6). NOTE 1 – The delay of the codec should be that specified by ITU-T for the common generic voice bearer and if there are any proposals for optional codecs include the information about those also. (For the satellite component, the satellite propagation delay is not included.)	Q	G2	A1.3.7.1 A1.3.7.2
A3.3.3	Handover/ALT quality Intra switch/controller handover directly affects voice service quality. Handover performance, minimum break duration, and average number of handovers are key issues.	Q	G2	A1.2.24 A1.2.24.1 A1.2.24.2 A1.4.6.1
A3.3.4	Handover quality for data There should be a quantitative evaluation of the effect on data performance of handover.	Q	G3	A1.2.24 A1.2.24.1 A1.2.24.2 A1.4.6.1
A3.3.5	Maximum user bit rate for data (bit/s) A higher user bit rate potentially provides higher data service quality (such as high quality video service) from the user's point of view.	Q	G1	A1.3.3
A3.3.6	Channel aggregation to achieve higher user bit There should also be a qualitative evaluation of the method used to aggregate channels to provide higher bit rate services.	q	G4	A1.2.32
A3.3.7	Voice quality Recommendation ITU-R M.1079 specifies that FPLMTS speech quality without errors should be equivalent to ITU-T Recommendation G.726 (32 kbit/s ADPCM) with desired performance at ITU-T Recommendation G.711 (64 kbit/s PCM). NOTE 1 – Voice quality equivalent to ITU-T Recommendation G.726 error free with no more than a 0.5 degradation in MOS in the presence of 3% frame erasures might be a requirement.	Q and q	G1	A1.2.19 A1.3.8
A3.3.8	System overload performance (not applicable to satellite) Evaluate the effect on system blocking and quality performance on both the primary and adjacent cells during an overload condition, at e.g. 125%, 150%, 175%, 200%. Also evaluate any other effects of an overload condition.	Q and q	G3	A1.3.9.1

A3.4	Flexibility of radio technologies			
A3.4.1	Services aspects			
A3.4.1.1	Variable user bit rate capabilities Variable user bit rate applications can consist of the following: – adaptive signal coding as a function of RF signal quality; – adaptive voice coder rate as a function of traffic loading as long as ITU-T Recommendation G.726 performance is met; – variable data rate as a function of user application; – variable voice/data channel utilization as a function of traffic mix requirements. Some important aspects which should be investigated are as follows: – how is variable bit rate supported? – what are the limitations? Supporting technical information should be provided such as – the range of possible data rates, – the rate of changes (ms).	q and Q	G2	A1.2.18 A1.2.18.1
A3.4.1.2	Maximum tolerable Doppler shift, F_d (Hz) for which voice and data quality requirements are met (terrestrial only) Supporting technical information: F_d	q and Q	G3	A1.3.1.4
A3.4.1.3	Doppler compensation method (satellite component only) What is the Doppler compensation method and residual Doppler shift after compensation?	Q and q	G3	A1.3.2.2
A3.4.1.4	How the maximum tolerable delay spread of the proposed technology impact the flexibility (e.g., ability to cope with very high mobile speed)?	q	G3	A1.3.1.3 A1.2.14 A1.2.14.1 A1.2.14.2 A1.3.10
A3.4.1.5	Maximum user information bit rate, R_u (kbit/s) How flexibly services can be offered to customers ? What is the limitation in number of users for each particular service? (e.g. no more than two simultaneous 2 Mbit/s users)	Q and q	G2	A1.3.3 A1.3.1.5.2 A1.2.31 A1.2.32
A3.4.1.6	Multiple vocoder rate capability – bit rate variability, – delay variability, – error protection variability.	Q and q	G3	A1.2.19 A1.2.19.1 A1.2.7
A3.4.1.7	Multimedia capabilities The proponents should describe how multimedia services are handled. The following items should be evaluated: – possible limitations (in data rates, number of bearers), – ability to allocate extra bearers during of the communication, – constraints for handover.	Q and q	G1	A1.2.21 A1.2.20 A1.3.1.5.2 A1.2.18 A1.2.24 A1.2.30 A1.2.30.1
A3.4.2	Planning			
A3.4.2.1	Spectrum related matters			

A3.4.2.1.1	Flexibility in the use of the frequency band The proponents should provide the necessary information related to this topic (e.g., allocation of sub-carriers with no constraints, handling of asymmetric services, usage of non-paired band).	q	G1	A1.2.1 A1.2.2 A1.2.2.1 A1.2.3 A1.2.5.1
A3.4.2.1.2	Spectrum sharing capabilities The proponent should indicate how global spectrum allocation can be shared between operators in the same region. The following aspects may be detailed: – means for spectrum sharing between operators in the same region, – guardband between operators in case of fixed sharing.	q and Q	G4	A1.2.26
A3.4.2.1.3	Minimum frequency band necessary to operate the system in good conditions Supporting technical information: – impact of the frequency reuse pattern, – bandwidth necessary to carry high peak data rate.	Q and q	G1	A1.2.1 A1.4.15 A1.2.5
A3.4.2.2	Radio resource planning			
A3.4.2.2.1	Allocation of radio resources The proponents and evaluators should focus on the requirements and constraints imposed by the proposed technology. More particularly, the following aspects should be considered: – what are the methods used to make the allocation and planning of radio resources flexible? – what are the impacts on the network side (e.g. synchronization of BSs, signalling,)? – other aspects. Examples of functions or type of planning required which may be supported by the proposed technology: – DCA, – frequency hopping, – code planning, – time planning, – nterleaved frequency planning. NOTE 1 – The use of the second adjacent channel instead of the adjacent channel at a neighbouring cluster cell is called “interleaved frequency planning”. In some cases, no particular functions are necessary (e.g. frequency reuse = 1).	q	G2	A1.2.25 A1.2.27 A1.4.15
A3.4.2.2.2	Adaptability to adapt to different and/or time varying conditions (e.g., propagation, traffic) How the proposed technology cope with varying propagation and/or traffic conditions? Examples of adaptive functions which may be supported by the proposed technology: – DCA, – link adaptation, – fast power control, – adaptation to large delay spreads. Some adaptivity aspects may be inherent to the RTT.	q	G2	A1.3.10 A1.2.27 A1.2.22 A1.2.14
A3.4.2.3	Mixed cell architecture (not applicable to satellite component)			

A3.4.2.3.1	Frequency management between different layers What kind of planning is required to manage frequencies between the different layers? e.g. – fixed separation, – dynamic separation, – possibility to use the same frequencies between different layers. Possible supporting technical information: – guard band.	q and Q	G1	A1.2.28 A1.4.15
A3.4.2.3.2	User adaptation to the environment What are the constraints to the management of users between the different cell layers? e.g. – constraints for handover between different layers, – adaptation to the cell layers depending on services, mobile speed, mobile power.	q	G2	A1.2.28 A1.3.10
A3.4.2.4	Fixed-wireless access			
A3.4.2.4.1	The proponents should indicate how well its technology is suited for operation in the fixed wireless access environment. Areas which would need evaluation include (not applicable to satellite component): – ability to deploy small BSs easily, – use of repeaters, – use of large cells, – ability to support fixed and mobile users within a cell, – network and signalling simplification.	q	G4	A1.1.3 A1.3.5 A1.4.17 A1.4.7 A1.4.7.1
A3.4.2.4.2	Possible use of adaptive antennas (how well suited is the technology) (not applicable to satellite component) Is RTT suited to introduce adaptive antennas? Explain the reason if it is.	q	G4	A1.3.6
A3.4.2.4.3	Existing system migration capability	q	G1	A1.4.16
A3.5	Implication on network interface			
A3.5.1	Examine the synchronization requirements with respect to the network interfaces. *Best case*: no special accommodation necessary to provide synchronization. *Worst case*: special accommodation for synchronization is required, e.g. additional equipment at BS or special consideration for facilities.	q	G4	A1.4.3
A3.5.2	Examine the RTTs ability to minimize the network infrastructure involvement in cell handover. *Best case*: neither PSTN/ISDN nor mobile switch involvement in handover. *Worst case*: landline network involvement essential for handover.	q	G3	A1.2.24 A1.4.6.1
A3.5.3	Landline feature transparency			
A3.5.3.1	Examine the network modifications required for the RTT to pass the standard set of ISDN bearer services. *Best case*: no modifications required. *Worst case*: substantial modification required, such as interworking functions.	q	G1	A1.4.7.1

A3.5.3.2	Examine the extent of the PSTN/ISDN involvement in switching functionality. *Best case*: all switching of calls is handled by the PSTN/ISDN. *Worst case*: a separate mobile switch is required.	q	G2	A1.4.6 A1.4.8
A3.5.3.3	Examine the depth and duration of fading that would result in a dropped call to the PSTN/ISDN network. The robustness of an RTTs ability to minimize dropped calls could be provided by techniques such as transparent reconnect.	Q and q	G3	A1.2.24 A1.4.14
A3.5.3.4	Examine the quantity and type of network interfaces necessary for the RTT based on the deployment model used for spectrum and coverage efficiencies. The assessment should include those connections necessary for traffic, signalling and control as well as any special requirements, such as soft handover or simulcast.	Q	G2	A1.2.30 A1.2.30.1 A1.4.9
A3.6	Handportable performance optimization capability			
A3.6.1	Isolation between transmitter and receiver Isolation between transmitter and receiver has an impact on the size and weight of the handportable.	Q	G2	A1.2.2 A1.2.2.1 A1.2.2.2
A3.6.2	Average terminal power output P_0 (mW) Lower power gives longer battery life and greater operating time.	Q	G2	A1.2.16.1.2
A3.6.3	System round trip delay impacts the amount of acoustical isolation required between hand portable microphone and speaker components and, as such, the physical size and mechanical design of the subscriber unit. NOTE 1 – The delay of the codec should be that specified by ITU-T for the common generic voice bearer and if there are any proposals for optional codecs include the information about those also. (For the satellite component, the satellite propagation delay is not included.)	Q and q	G2	A1.3.7 A1.3.7.1 A1.3.7.2 A1.3.7.3
A3.6.4	Peak transmission power	Q	G1	A1.2.16.1.1
A3.6.5	Power control characteristics Does the proposed RTT utilize transmitter power control? If so, is it used in both forward and reverse links? State the power control range, step size (dB) and required accuracy, number of possible step sizes and number of power controls per second, which are concerned with mobile station technology complexity.			
A3.6.5.1	Power control dynamic range Larger power control dynamic range gives longer battery life and greater operating time.	Q	G3	A1.2.22 A1.2.22.3 A1.2.22.4
A3.6.5.2	Power control step size, accuracy and speed	Q	G3	A1.2.22 A1.2.22.1 A1.2.22.2 A1.2.22.5
A3.6.6	Linear transmitter requirements	q	G3	A1.4.10
A3.6.7	Linear receiver requirements (not applicable to satellite)	q	G3	A1.4.11
A3.6.8	Dynamic range of receiver The lower the dynamic range requirement, the lower the complexity and ease of design implementation.	Q	G3	A1.4.12
A3.6.9	Diversity schemes Diversity has an impact on hand portable complexity and size. If utilized describe the type of diversity and address the following two attributes.	Q and q	G1	A1.2.23 A1.2.23.1 A1.2.23.2

A3.6.10	The number of antennas	Q	G1	A1.2.23.1
A3.6.11	The number of receivers	Q	G1	A1.2.23.1
A3.6.12	Frequency stability Tight frequency stability requirements contribute to handportable complexity.	Q	G3	A1.4.1.2
A3.6.13	The ratio of "off (sleep)" time to "on" time	Q	G1	A1.2.29 A1.2.29.1
A3.6.14	Frequency generator step size, switched speed and frequency range Tight step size, switch speed and wide frequency range contribute to handportable complexity. Conversely, they increase RTT flexibility.	Q	G2	A1.4.5
A3.6.15	Digital signal processing requirements Digital signal processing can be a significant proportion of the hardware for some radio interface proposals. It can contribute to the cost, size, weight and power consumption of the BS and influence secondary factors such as heat management and reliability. Any digital circuitry associated with the network interfaces should not be included. However any special requirements for interfacing with these functions should be included. This section of the evaluation should analyse the detailed description of the digital signal processing requirements, including performance characteristics, architecture and algorithms, in order to estimate the impact on complexity of the BSs. At a minimum the evaluation should review the signal processing estimates (MOPS, memory requirements, gate counts) required for demodulation, equalization, channel coding, error correction, diversity processing (including Rake receivers), adaptive antenna array processing, modulation, A-D and D-A converters and multiplexing as well as some IF and baseband filtering. For new technologies, there may be additional or alternative requirements (such as FFTs). Although specific implementations are likely to vary, good sample descriptions should allow the relative cost, complexity and power consumption to be compared for the candidate RTTs, as well as the size and the weight of the circuitry. The descriptions should allow the evaluators to verify the signal processing requirement metrics, such as MOPS, memory and gate count, provided by the RTT proponent.	Q and q	G1	A1.4.13
A3.7	Coverage/power efficiency			
A3.7.1	Terrestrial Coverage efficiency: – the coverage efficiency is considered for the lowest traffic loadings; – the base site coverage efficiency can be quantitatively determined by addressing coverage limitation and/or by calculating the maximum coverage range for the lowest traffic loading.			
A3.7.1.1	Base site coverage efficiency The number of base sites required to provide coverage at system start-up and ongoing traffic growth significantly impacts cost. From § 1.3.2 of Annex 2, determine the coverage efficiency, C (km^2/base sites), for the lowest traffic loadings. Proponent has to indicate the background of the calculation and also to indicate the maximum coverage range.	Q	G1	A1.3.1.7 A1.3.1.7.1 A1.3.1.7.2 A1.3.4

A3.7.1.2	Method to increase the coverage efficiency Proponent describes the technique adopted to increase the coverage efficiency and drawbacks. Remote antenna systems can be used to economically extend vehicular coverage to low traffic density areas. RTT link budget, propagation delay system noise and diversity strategies can be impacted by their use. Distributed antenna designs – similar to remote antenna systems – interconnect multiple antennas to a single radio port via broadband lines. However, their application is not necessary limited to providing coverage, but can also be used to economically provide continuous building coverage for pedestrian applications. System synchronization, delay spread, and noise performance can be impacted by their use.	q	G1	A1.3.5 A1.3.6
A3.7.2	Satellite Normalized power efficiency Supported information bit rate per required carrier power-to-noise density ratio for the given channel performance under the given interference conditions for voice Supported information bit rate per required carrier power-to-noise density ratio for the given channel performance under the given interference conditions for voice plus data mixed traffic.	Q	G1	A1.3.2.4 A1.3.2.4.1 A1.3.2.4.2