WiMAX® VoIP Solutions for 4G Networks

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WiMAX® VoIP Solution for 4G Networks

Executive Summary

This white paper presents an overview on how to deliver PSTN quality VoIP services over WiMAX® networks.

Many service providers, including fixed and mobile operators, recognize the indisputable business case for deploying VoIP. The ubiquity of IP as a networking technology at the customer premises opens the possibility of deploying a vast range of innovative converged multimedia services that simply cannot be cost effectively supported over today's PSTN infrastructure, especially when PSTN network equipment is aging and the replacement costs tend to be more expensive than deploying IP networks.

On the other hand, many users have been enjoying the reliable high-quality PSTN voice services for many years. In order for the VoIP services to be successfully deployed, a PSTN quality end-to-end VoIP solution is required.

In order to enable a PSTN quality VoIP service over WiMAX networks, the following considerations and network solutions are necessary:

- An equivalent PSTN service set offering for WiMAX networks
- WiMAX VoIP ecosystem that provides all the required network elements from terminal to VoIP server
- Network architecture that has been supported and developed by the WiMAX Forum® to support the VoIP services
- A high quality of service to maintain a similar user experience as PSTN
- Key Performance Indicators (KPIs)
- Wireless and wired bandwidth efficiency to deploy VoIP over WiMAX to maximize the profit margin for Fixed/Mobile Convergent (FMC) operators
- Signaling protocols that enable time-to-market opportunity and rich service set support
- An architecture and solution which meets regulatory requirements such as Lawful Intercept and Emergency Services to enable public carriers for VoIP services.
- For Mobile WiMAX VoIP service, inter-RAT interworking/handover, and roaming support are required to provide ubiquitous voice services.

The audience for this white paper includes WiMAX network operators and service providers as well as their technical engineers. The white paper describes the technical challenges for supporting VoIP services over WiMAX networks as highlighted above, as well as technical recommendations on how to address those technical challenges in order to deploy PSTN quality VoIP services over WiMAX networks.

1 Introduction

Voice over IP (VoIP) is the transport of voice using the Internet Protocol (IP) to carry packetized voice data over different types of wired or wireless media (e.g. DSL, Cable, Ethernet, Wi-Fi, WiMAX® networks, etc.). However, VoIP over wireless medium is still in the infancy stage due to limited RF capacity and the challenges to maintain radio channel performance while maintaining a level of QoS to ensure a positive user experience.

WiMAX technology is the first fourth-generation mobile broadband technology enabling QoS support and providing high broadband capacity. It enables an attractive revenue opportunity for mobile and fixed network operators to capitalize the VoIP business case over the WiMAX broadband medium.

The purpose of this white paper is to present the WiMAX network solutions, to provide best practice guidelines that aggregate existing standards and recommendations to WiMAX network operators on deploying VoIP.

This white paper outlines the VoIP deployment scenarios, system and regulatory requirements, ecosystem, technical challenges, key performance expectations, network design considerations, and WiMAX standardized solutions that enable VoIP support over WiMAX networks.

The white paper is organized as follows to describe how WiMAX networks can provide PSTN quality VoIP services:

- VoIP deployment scenarios and opportunity for WiMAX networks
- Scope of the this white paper on the technical discussions is focused on
- Key requirements for VoIP over WiMAX networks
- The ecosystem regarding the solution space
- The VoIP network architecture options and network components for WiMAX operations as well choices of protocols
  - IMS-based VoIP over WiMAX networks
  - Non-IMS based VoIP over WiMAX networks
- Design considerations and network solution for VoIP services over WiMAX networks
  - QoS and bandwidth management
    - Performance target (delay, jitter and packet loss)
    - Policy control (e.g. PCC support)
    - Bearer control
      - IP backhaul (Diffserv)
      - Airlink scheduling and radio resource management
        - Airlink scheduling class selection (e.g. UGS vs. ert-PS etc.)
        - HARQ to minimize packet loss
    - Header Compression
  - Reliability and availability
  - Regulatory Issues
    - Lawful Interception
    - Emergency call support
    - Location Information
  - Call Routing and numbering plans
  - Accounting Support (Postpaid and Prepaid)
  - Network Interoperability, Interworking and Roaming Support
  - Device Management support for remote auto-configuration and service provisioning
- Conclusions

2 VoIP Network Deployment Scenarios

Several categories of VoIP Networks are to be considered based on the terminal and the location of the VoIP platform[1] as follows:

1. Fixed User End-User VoIP Termination– VoIP platform located on the same premises as the WiMAX® Provider's network.

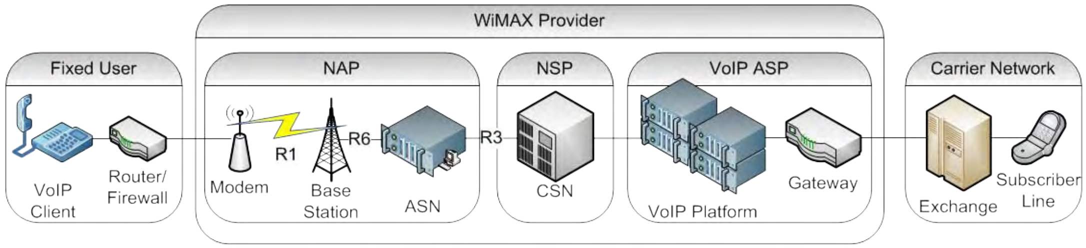


Figure 2-1: WiMAX® VoIP Network Deployment Scenario#1

2. Mobile User – VoIP platform located on the same premises as the WiMAX Provider's network

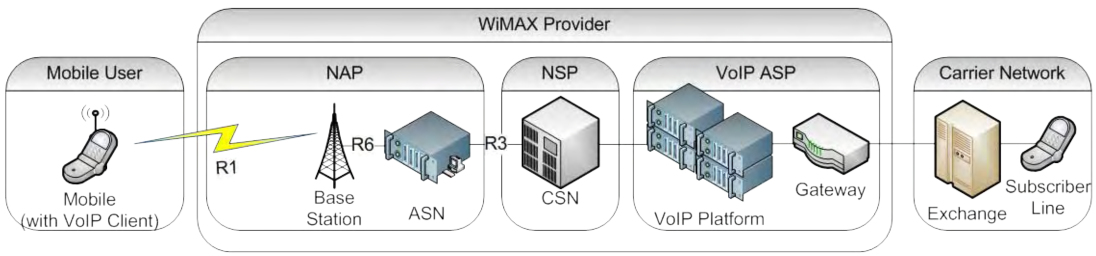


Figure 2-2: WiMAX® VoIP Network Deployment Scenario#2

3. Fixed User – VoIP platform located at a separate location from the WiMAX Provider's network.

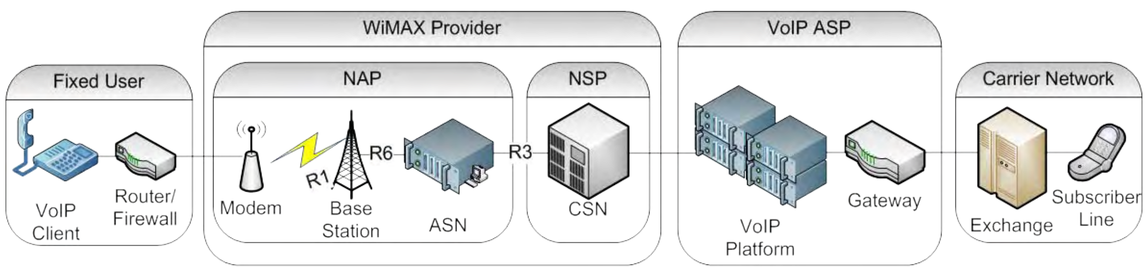


Figure 2-3 : WiMAX® VoIP Network Deployment Scenario#3

4. Mobile User – VoIP platform located at a separate location from the WiMAX Provider's network

[1] The "VoIP Platform" here is to refer the group of network elements such as the Session Border Controller (SBC), Soft-switch, Feature Server, Media Server, Provisioning Server that are required to provide the management, control and bearer control functions for VoIP support. More details on these set of functions are discussed in section 3.3.

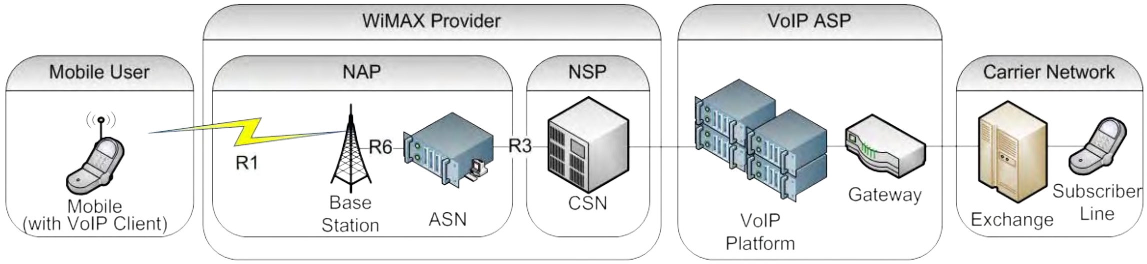
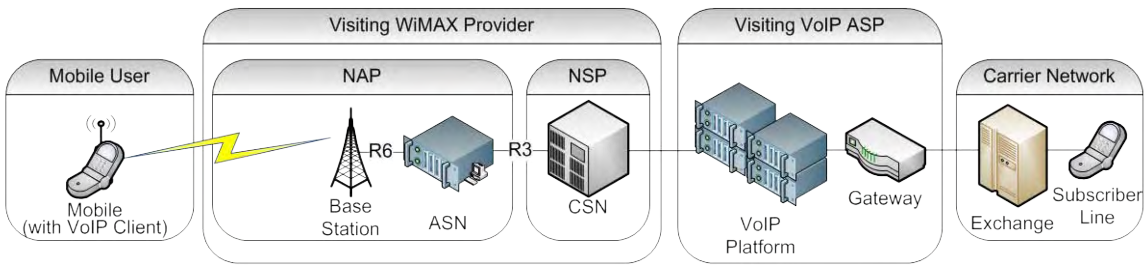


Figure 2-4 : WiMAX® VoIP Network Deployment Scenario#4

5. Roaming – When a WiMAX user is roaming the VoIP platform used may be located either in the visited network or the home network. The depiction below shows both cases.



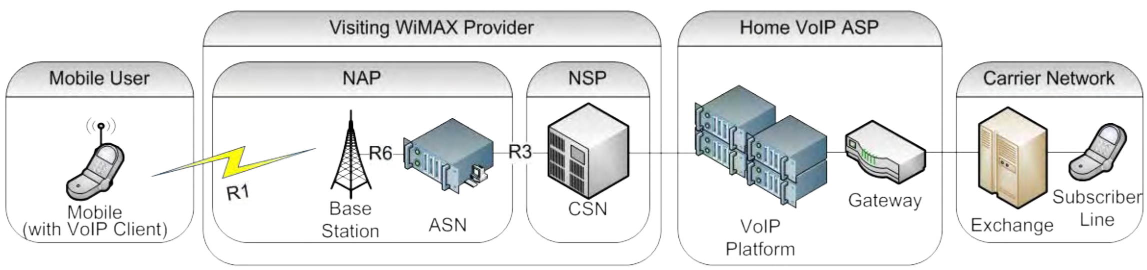


Figure 2-5 : WiMAX® VoIP Network Deployment Scenario#5

3 VoIP Ecosystem

An ecosystem is a unit of interdependent systems interacting as a functional whole. The purpose of the interdependent systems is to make and receive calls. The four logical networks that make up the VoIP Ecosystem include the following:

- **PSTN Carrier Network**: Made up of traditional Cell and POTS line providers interconnecting using a circuit switched network.
- **VoIP Application Service Provider Network**: The network where the VoIP equipment is located.
- **WiMAX® Provider Network**: For the objective of this paper, this network will include the WiMAX base stations, ASN and CSN as depicted below.

- **End-User VoIP Termination**: The End-User VoIP Termination that hosts the VoIP client and other network elements, not including the WiMAX modem.

Not every implementation is the same and some of these networks may overlap, the following diagram depicts the logical functions of the VoIP ecosystem with respect to non-IMS based VoIP over WiMAX networks (see section 6.1 for details).

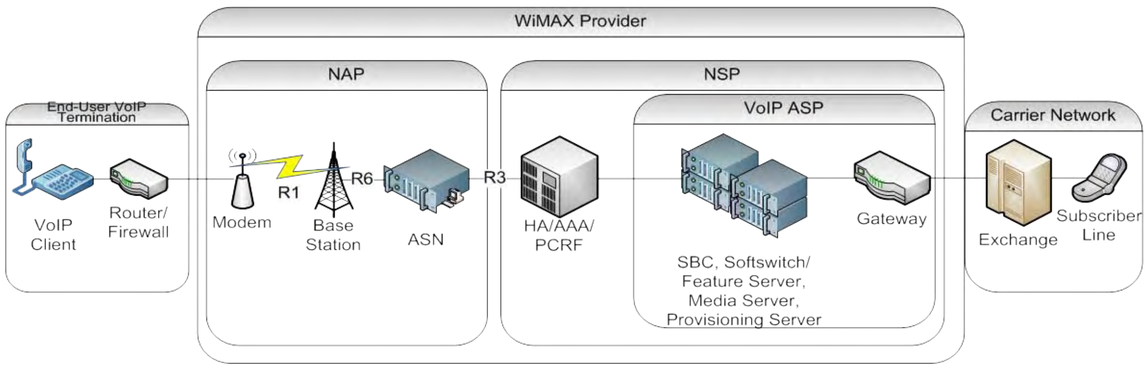


Figure 3-1 : Simple Ecosystem

This section will provide a high level explanation of the elements in each network, the logical function of those elements as it relates to VoIP, and concepts to consider concerning quality. The reality is that every network is different with some being more complex than others. Some hardware vendors have combined multiple logical network functions into a single piece of hardware. This white paper will not attempt to provide an exhaustive list of scenarios and will use separate elements to represent different logical functions as much as possible.

3.1 End-User VoIP Termination

The end user network can include several logical functions that are described below. The logical functions include a VoIP Client and WiMAX router/firewall.

3.1.1 VoIP Client

The VoIP Client used by the end-user to make and receive calls is the front door to the VoIP service and should be selected carefully based on quality and end-user needs. A VoIP Client is a logical function that enables VoIP services. Below are examples of VoIP Clients.

- **ATA (Analog Telephony Adaptors)**: A device that converts the analog signals from a conventional phone into a format acceptable for transmission over an internet connection. Typically used in residential VoIP offerings allowing the end-user to utilize their existing analog phone. In the WiMAX case, typically an ATA connects a WiMAX CPE to a conventional phone. The WiMAX CPE connects with a WiMAX BS to enable VoIP services.
- **IP Phones**: A telephone with built-in IP signaling protocol (SIP) that converts voice into IP packets and vice versa. Typically used in Small/Medium Business offerings. The phones typically also have buttons dedicated to business features such as hold, transfer, voicemail, etc.
- **Soft phones**: A software program that runs on a PC (desktop or laptop or smart-phone) that allows calls to be made and received over the Internet (VoIP). A headset, or a microphone and

speakers, is typically used in place of a telephone. Typically used as a primary device by those on the road often or a supplemental device for a residential or SMB user.

Provisioning of these VoIP clients will be discussed section 3.3.3 below.

3.1.2 Firewall

Not all End-user networks will have a router or firewall but it is important to understand its affect on SIP if they are present.

The firewall is designed to prevent unauthorized access. Such methods include layer 2-4 filters such as MAC, source/destination IP address, protocols, ports, etc. The firewall must be configured to allow incoming/outgoing VoIP calls.

3.1.3 Router

The router is designed to provide additional addressing for internal hosts behind the WiMAX modem by assigning a private IP address (RFC 1918 [16]). Network Address Translation (NAT RFC 3022 [18]) allows multiple hosts behind the router to utilize a single public IP address for outbound connections but poses a challenge for identifying specific clients on incoming connections. Wi-Fi with router capability is often integrated into WiMAX modems to extend the indoor coverage.

Because the nature of a voice application is to make and receive calls, and calls can be received from different end-points, routers/firewalls pose a challenge to VoIP. One solution to the challenge of routers is addressed in the Section 3.3.1 on Session Boarder Controller below.

3.2 WiMAX® Provider

For the purpose of this paper, the WiMAX Provider network consists of the Network Access Provider (NAP) and the Network Service Provider (NSP), as defined by the WiMAX Forum network reference model [13]. Note that the WiMAX Modem is managed by the WiMAX provider.

3.3 VoIP ASP Network

The VoIP Application Service Provider (ASP) network is simply the network that hosts the VoIP service elements. This could be hosted on the WiMAX Provider's network or on a separate network. The functional elements of a VoIP platform are described as follows:

3.3.1 Session Border Controller (SBC)

The term, Session Border Control (SBC), is not a standardized set of functions. Instead, SBC has evolved to address the wide range of issues that arise when voice and multimedia services are overlaid on an IP infrastructure. These include:

- security and prevention of service abuse to ensure Quality of Service (QoS);
- maintaining privacy of carrier and user information;
- resolution of VoIP protocol problems arising from the widespread use of firewalls and network address translation (NAT), and the vast array of differing protocols and dialects used in VoIP networks.

The Session Border Controller (SBC) performs the functions of analyzing and altering signaling between clients to ensure protocol compatibility, security, and control across IP network boundaries. Some challenges introduced with VoIP addressed by the SBC include the following: Incompatible SIP clients, (as mentioned in Section 3.1.3) clients located behind a router, and security. The functions of SBC as discussed in this white paper apply to both IMS and non-IMS system.

3.3.2 Softswitch/Feature Server/IMS

The hub of the VoIP platform is the Softswitch/Feature Server/IMS. It possesses the logic to manage the registration of provisioned clients, the setup and disconnect of simple calls as well as all the logic to perform the complex features required by both residential and business users such as forward, transfer and call waiting.

3.3.3 Provisioning Server

Provisioning is the process of setting configuration parameters. The provisioning server hosts the parameters generic for each client type and those specific to each user. The VoIP client provisioning component enables the client to be remotely configured automatically which in turn enables End-users to simply "plug-and-play". WiMAX provisioning methods are OMA-DM [14] or TR-69 [15].

Some VoIP clients are integrated with the WiMAX modem and therefore could use common provisioning methods. The WiMAX Forum® recommends using OMA-DM [13] and TR-69 [14].

3.3.4 Media Server

The Media Server is the element responsible for playing RTP messages like Interactive Voice Response (IVR) voice prompts and voice messages as the Softswitch/Application Server directs.

3.3.5 Billing Support, AAA and Online Accounting Servers

The Billing Support system enables the Service Provider to define the rates and charges associated with the products they create for the End-User. It tracks individual Call Detail Records (CDR's), Online and Offline Charging, Monthly Recurring Charges (MRC's), Non-Recurring Charges (NRC's), and other Transactions.

AAA (Authentication, Authorization and Accounting)/HSS (Home Subscriber Server) as referred here is to support the Authentication piece. The AAA and the online/offline Servers provide the accounting records required for the billing systems.

The authorization policies for a voice application can be different from the policies for data; however, since the voice call rides on the data layer, if the data is not authorized, the voice call will not work.

3.3.6 Gateways

A Gateway (or Media Gateway) is a device used to translate media streams between different technologies. With VoIP, the gateway converts the media stream as well as the control protocol between packet-switched and circuit-switched methods.

3.4 PSTN Carrier Network

The Public Switched Telephone Network (PSTN) is the traditional circuit-switched telephone network made up of fixed Plain Old Telephone Service (POTS) and mobile lines. The PSTN is made up of interconnected switches or Exchanges that ultimately terminate to Subscriber Lines. The ITU-T is the standards board that governs this network.

4 Quality of Service (QoS)

As the quantity and variety of traffic being processed over a WiMAX®-based network increases, so does the need for implementing comprehensive Quality of Service (QoS) technologies to ensure the optimal Quality of Experience (QoE) for the end-user. This is especially true when supporting real-time applications like Voice over IP (VoIP).

In addition to the challenges faced by traditional IP-based networks, the WiMAX-based network also faces the challenges of dynamic radio environment. The purpose of this section is to outline the specific challenges faced by WiMAX operators seeking to optimize the QoE for VoIP services and applications.

4.1 WiMAX® Quality of Service

QoE is a subjective measure of the end user's experience of the service and application. It is based on the end-to-end performance of the entire network, not just the quality of the individual IP or WiMAX links. Therefore, in order to maximize the QoE, comprehensive QoS solutions must be implemented across every segments of the network (i.e. over wired or wireless transport) and VoIP components (e.g. codec, jitter buffer etc.), where over-engineering/provisioning the backhaul transport and the control elements (e.g. SIP server, AAA server etc.) along with careful network planning and intelligent radio resource management and airlink scheduling design are needed to operate within the limited spectrum. More on the WiMAX specific network solution and technology will be discussed later in this white paper.

Important E2E network QoS challenges to support VoIP over WiMAX networks include:

- The end-to-end network latency, packet loss and jitter.
- QoS issues inherent to wireless technologies such as WiMAX technology.

These two challenges are explained in more detail below and recommendations are included for overcoming these challenges to provide a high-quality VoIP call experience.

4.2 End-to-End Network QoS Challenges

The following sections explain the inherent QoS challenges to providing quality VoIP service at the IP layer and above. Because WiMAX technology is also part of the end-to-end IP network, it also has to deal with the QoS challenges that are highlighted in this section.

4.2.1 Latency

Latency is the end-to-end delay of the voice signal from the person on one side of the conversation to the person on the other end of the conversation. It consists of three primary factors: network delay, playout delay, and CODEC delay.

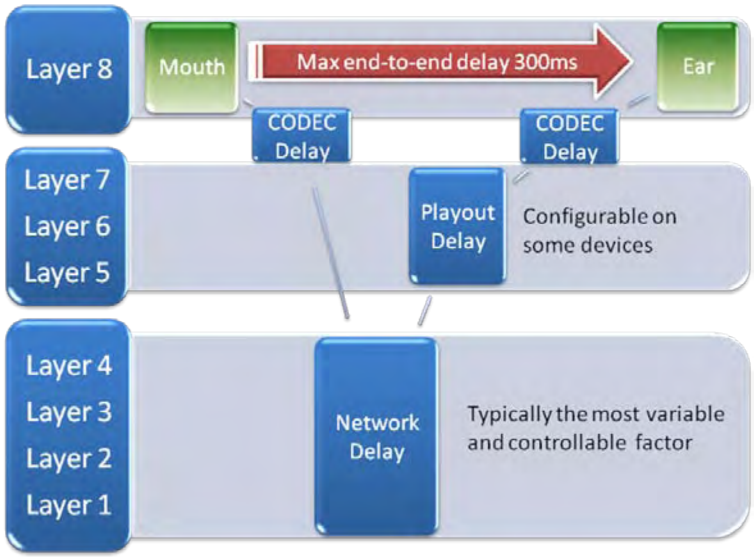


Figure 4-1: Delay Waterfall Diagram

- Network Delay

The term *latency* is often used when referring to network delay even though network delay is only one of three latency factors. The reason for this is that the network delay is typically the most variable and controllable of the three. Network delay consists of the time it takes the IP packet to traverse from endpoint to endpoint. This can be influenced by network congestion, routing changes, geographic separation of the endpoints, etc.

As network delay increases, the call quality is increasingly affected. To provide best voice quality, the one way end-to-end (mouth to ear) latency is preferred to be 150 ms or less. Anything higher than 150 ms will begin to affect call quality. Anything above 300 ms is generally considered unacceptable quality. It is recommended that the Service Provider implements either Differentiated Services (“DiffServ”) [53][54] or Integrated Services (“IntServ”) [55][54] in order to control the network delay.

For example, IntServ implements the parameterized approach in which applications request and reserve resources using a set of QoS parameters, e.g. minimum tolerable rate, maximum delay, tolerated jitter through the network. DiffServ implements the prioritized model, in which packets are marked and classified according to the type of service they need, and in response to these markings, routers and switches use various queuing strategies to tailor performance to requirements. WiMAX networks are capable of supporting both approaches.

- Playout Delay

Playout delay is the delay caused by the voice packets waiting in the jitter buffer. At the receiving end of each stream of digital voice packets, the CODEC must decode the packets from digital back to analog. However, the internet and network traffic arrival rate in general is not constant. One packet may take 100 ms to arrive and the next packet may take 120 ms. This variance is known as jitter. The jitter buffer shall be no more than an artificial delay built into the CODEC in order to compensate for the possible delay in packets. However, an increase in the size of the jitter buffer also increases the E2E delay budget of the call.

- CODEC Delay

CODEC delay is the delay caused by the encoding and decoding process of a CODEC. A CODEC is a coder and decoder of analog and digital packets. The algorithm used to digitize the analog signal introduces a certain amount of CODEC-specific delay. The process of digitization also takes an amount of processing time of the signal and is dependent upon the hardware/software and CODEC in use.

A raw, uncompressed voice call will consume significantly more bandwidth than a call using an efficient CODEC. A number of different CODECs are available for VoIP, but the two primary CODECs used by most VoIP devices and networks are G.711 [49][50] and G.729 [51]. Both CODECs are widely supported, but it is recommended that Service Providers use the G.729 [51] CODEC because it provides the highest voice-quality-to-bandwidth-usage ratio and more optimal for human speech pattern whereas G.711 [49][50] is more optimal for constant bit-rate real-time traffic such as fax. One thing to note is that G.729 [51] is a royalty based CODEC.

4.2.2 Packet Loss

VoIP requires an IP-based network where each voice conversation must be packetized (broken up into small data packets). Packet loss occurs when one or more of these packets are lost during transmission. Significant packet loss could have a detrimental effect on the end-user's QoE. Packet loss can occur due to network congestion or connectivity errors between endpoints. The recommend packet loss for VoIP should be no more than 1%.

To decrease packet loss, the Service Provider should ensure that sufficient bandwidth is available for the given VoIP session regardless of whether it is over a wired or wireless medium.

4.2.3 Jitter

Jitter is variation in the order and time in which packets are sent and received. Packets are numbered and sent chronologically as they are created. Due to network congestion or the ability of packets to take different routes over the network, packets may arrive out of order or with varying delays. In order to maintain a high QoE for the end-user, voice packets must arrive in a fairly consistent manner.

In an attempt to "smooth" the incoming voice packets, a jitter buffer has been implemented in all modern VoIP deployments. The jitter buffer holds a small reserve of voice data on the receiving end and releases the data in a consistent manner.

In order to ensure good call quality, it is recommended that the jitter be less than 50 ms, although 20 ms is preferred.

4.3 WiMAX®-Specific QoS Challenges

4.3.1 Wireless QoS Challenges

In comparison to a wired broadband technology, the wireless broadband WiMAX technology, will encounter more intense interference and noise over its wireless transport. A well-engineered wired technology provides the end-point a more deterministic performance over a controlled environment. Wireless technology, however, are less predictable because of dynamic changes that can weaken or distort the signal, and therefore increase packet loss, latency, or jitter.

The wireless channel is constrained by limited spectrum and device power. The system design strategy for constant resource allocation to meet all VoIP users is quite different from a wired medium. A wireless network requires more advance planning and real-time airlink performance monitoring as well as concise algorithms to provide adaptive radio resource allocation, not only at the system level but also at the device level, while supporting many other users who may use other type of services and applications, such as file transfer.

Common challenges to WiMAX QoS include packet loss over the wireless channel due to decay of signal over distance, obstructions in line of sight, interference and noise, handover, increase of end-to-end latency and jitter due to congestion and contention among multiple users and service flows, etc.

The issues described above are not unique to WiMAX technology. The differentiation that WiMAX technology offers to enable a high quality successful VoIP service deployment compared to prior 3G technology is its native design with an adaptive high capacity airlink protocol and a network architecture to support more stringent QoS parameters and efficient radio resource allocation.

More on the technical recommendation on the WiMAX radio resource management will be discussed in a later section of this white paper.

4.3.2 Integrated WiMAX® and IP-Based QoS Challenges

In addition to the wireless QoS challenges, there is the important consideration of how to integrate end-to-end QoS between the WiMAX airlink and the IP networks as their QoS protocol design strategies are not the same.

Although WiMAX specifications provide the vital build blocks for supporting E2E QoS, the system integration and the network engineer strategy are dependent on the operator's decision on the network deployment policy.

4.4 WiMAX®-Specific QoS Recommendations

WiMAX technology provides a robust foundation for QoS with multiple hardware and software improvements over traditional QoS. It is important to select the WiMAX base station and subscriber stations as well as the backhaul transport equipment together with WiMAX ASN-GW which are responsible for the transition from the WiMAX core network (ASN) to the IP backbone and vice versa. The following sections detail general and specific WiMAX recommendations that should be considered when designing a WiMAX network for high-quality VoIP.

4.4.1 Service Level Agreement

A service provider that wants to guarantee QoS for VoIP services can offer differentiated services with specific Service Level Agreements (SLAs) and charge for the services accordingly. Only the user who has subscribed to a particular QoS scheduling services as specified per IEEE Std 802.16-2009 [55], for example extended Real-time Polling Service (ert-PS) [55], will be assigned a high QoS for the voice session.

A Service Level Agreement to support VoIP services and applications should at least include support for prioritizing real-time VoIP traffic over other non real-time traffic, such as file transfer for example.

4.4.2 WiMAX Subscriber Station

There are three important considerations when selecting WiMAX devices to support VoIP.

- Verify the device supports WiMAX QoS scheduling services and the mapping of IEEE Std 802.16-2009 scheduling services [60] to service flows as required. The QoS scheduling services [60] that are optimized for VoIP and should be considered are Unsolicited Grand Service (UGS), Extended Real-time Service (ertPS), and Real-time Service (rtPS).
- Verify the device will work in the majority of configurations expected from end-users.

- Flexible service flow configuration rule customization must be allowed. It should be able to map services flows based on IP address, MAC, VLAN or port.

As explained earlier, WiMAX service flows provide the ability to map specific traffic to different parameters and priorities based either on traffic type or on various device or connection specific parameters like IP address, MAC address, VLAN tag or destination IP address. The WiMAX device classifies all traffic traversing towards the WiMAX network and may modify the traffic depending on its service flow mapping.

4.4.3 Network Architecture

An important consideration when planning for QoS is to ensure that the WiMAX and underlying IP network have been architected properly to ensure sufficient resources are available to each subscriber.

The number of subscribers per sector, overbooking ratios, etc. should be considered. The underlying core network must also have sufficient bandwidth for all network segments to ensure QoS classifications will be honored.

4.4.4 CODEC

The choice of CODEC used is important because it determines the required bandwidth per call. G.711 [49][50] and G.729 [51] are the most widely used codecs in existing networks, however, G.729 CODEC is widely used in wireless networks . G729 has a data rate of 8 kbps and a 10-30 ms sample period (depending on implementation). Average bandwidth usage is ~40 kbps per call. The table below described the G.729 codec performances.

The G.711 codec is currently used in a wide range of applications. Its voice sampling rate is 8 kHz and each sample is encoded with 8 bits resulting in a constant 64 kbps bit rate and offers very good voice quality. Samples can be packed into frames every 10 ms or another longer sampling rate.

The G.729 codec can also generate speech frames every 10 ms or longer sample rate .Each 10 ms frame contains 80 voice samples (collected at a sampling rate of 8000 samples per second), or another longer sampling rate. However, it requires a 5 ms look-ahead delay before producing any new frame. It is developed for multimedia simultaneous voice and data applications.

Table 4-1 : G.711 vs. G.729 CODEC Performance

Codec	G.711	G.729
Sample Time (ms)	10	10
Frame Size (bits)	640	80
Packets Per Second	100	100
IP, RTP, UDP headers (bits)	320	320
IP-CS data rate (Kbps)	96	40
Data Rate Saving Efficiency with IEEE 802.16 Packet Header Suppression (PHS) (see Note-1) (see later section 4.4.9 for more details for PHS)	20.4%	40.2%
Data Rate Saving Efficiency with Robust Header Compression (RFC 3095[18]) (see Note-1) (see later section 4.4.9 for more details for ROHC)	26.7%	52.7%

NOTE-1: *Courtesy of the following: "A Comparative Study of Bandwidth Requirements of VoIP Codecs Over WiMAX Access Networks – Ashraf A. Ali, S. Vassilaras, K. Ntagkounakis, 2009 Third International Conference on Next Generation Mobile Applications, Services and Technologies.*

Silence suppression is an effective way to reduce bandwidth usage of a voice call by not transmitting information or transmitting less information over the network when one of the parties involved in a voice call is not speaking. A CODEC with silence suppression capability should be used. Both G.711 and G.729 supports silence suppression.

4.4.5 Subscribers

An important consideration when addressing QoS is the number of subscribers per sector. Usually, adding VoIP to a network decreases the overall subscriber capacity due to VoIP's higher requirements on delay and jitter compared to other non-real-time applications.

Latency increases as the sector utilization and/or the number of subscribers increases. As usage increases, the limiting factor for the maximum number of subscribers in a given sector may not be the available sector bandwidth, rather, the limitation is the maximum acceptable packet latency to maintain a high quality voice conversation.

4.4.6 Polling Induced Latency

Although many things can contribute to latency, one of the primary causes is related to uplink polling and bandwidth requests. The only delay on the downlink is for scheduling, but on the uplink the mobile station must request and be granted bandwidth. Polling is a process that specified in IEEE Std 802.16airlink interface specification [56]. The WiMAX Base Station (BS) periodically allocates transmission opportunities to allow the Mobile Station (MS) to send a bandwidth request. If the amount of bandwidth needed at the MS is known in advance, the BS may use granting to avoid polling delay, in which the WiMAX BS periodically allocates resources for the MS to send its VoIP packets directly. After an MS inactivity period, an MS may "piggyback" bandwidth requests on the Channel Quality Indication Channel (CQICH). This avoids forcing the MS to wait for the BS to poll for the uplink resource allocation [56]. Periodic granting is particularly useful in uplink when the packet size and the uplink transmission schedule of a VoIP call are constant.

However, it is important that the number of registered subscribers in each sector be regularly monitored to ensure usage does not exceed capacity. When capacity limits are reached, actions should be taken to increase capacity or balance load.

WiMAX airlink scheduling and implementation recommendations are further described in a later section.

4.4.7 Packet Rate

A difference between wireless and wire line networks is the limitation of Packets Per Second (PPS) that a wireless link can process. Generally wireless networks can transmit fewer packets per second and therefore the packets that are transmitted must be optimized for traffic. An additional consideration is the existence of fixed control overhead, such as MAC headers, for each transmission. By increasing the packet payload (i.e., the size of the packet) and lowering the frequency at which packets are sent, wireless networks are able to manage the packet rate, and support more simultaneous VoIP users. It is recommended that for the G.729 CODEC, the WiMAX codec is configured at the sampled interval of 20 ms or 30 ms. Certainly, the longer sampling rate will have negative impact on the end-to-end delay budget.

4.4.8 Jitter Buffer

VoIP call quality can be significantly improved by selecting VoIP equipment with sufficiently large dynamic jitter buffers. High quality jitter buffers can greatly improve the quality of voice over a wireless network without requiring specific configuration change in the wireless network itself.

However, WiMAX network shall minimize the jitter because the jitter will have negative impact to the overall end-to-end delay budget for voice.

4.4.9 Header Compression

The packets delivered to OSI model layer 2 may have very large headers, sometimes as long as 120 bytes, as is the case for uncompressed RTP/UDP/IPv6 packets. Much of the header information does not change dynamically (e.g. protocol version, IP address, etc.) and hence, if it is known in advance, it may be suppressed on the transmitter side and recovered at the receiver side. IEEE Std 802.16 [56] defines the Payload Header Suppression (PHS) mechanism and protocol that exchanges PHS rules during WiMAX connection establishment. PHS rules are expressed in terms of a Payload Header Suppression Index (PHSI) which references a Payload Header Suppression Field (PHSF).

The transmitter performs packet classification and suppresses the parts of the payload header in the MAC SDU that match the predefined PHS Rule(s). The receiving entity restores the suppressed parts based on the PHS rule(s) exchanged during connection establishment. This allows PHS to compress all static packet header fields (e.g., in RTP/ UDP/ IP packets) down one byte (i.e., the PHSI info). The PHS rules are designed in such a way that fields of the header that do not change for the entire duration of the service flow are suppressed. Only changing fields are transmitted. The standard allows for multiple rules for every service flow to allow for multiple streams to be transmitted over a service flow.

There are also header compression algorithms that compress packet headers by other means than PHS. WiMAX supports ROHC (Robust Header Compression) (RFC 3095) [19] which includes a compression algorithm and an information exchange protocol, can compress IP/UDP/RTP headers to just over one byte and is robust even in the presence of severe channel degradation. Compared to other header suppression or compression algorithms, ROHC is sensitive to wireless airlink conditions and will adapt to the airlink conditions by adjusting the size of the compressed packets as well as compression information accordingly in order to minimize the packet loss probability over the error wireless link.

VoIP traffic is fairly periodic and the payload is small. As an example, consider the following calculation. Assume VoIP traffic model as in ITU G.279a with silence suppression. The source traffic rate is 8 kbps with a voice activity factor of 0.5. The voice frames are generated every 10 ms. Every 20 ms, 2 voice frames (amounting to 160 bits = 20 bytes) are aggregated into an IP packet and sent. For every voice payload transmitted over the air, 8 bytes of MAC overhead and at least 40 bytes of RTP/UDP/IP overhead are incurred. This results in a very low protocol efficiency of ~ 30% (=20/68). With header compression, this protocol efficiency increases to ~ 70% (=20/29), assuming that header compression results in a minimum of 1 byte compressed RTP/UDP/IP header (on average this may be 3-4 bytes). Thus, header compression significantly increases the efficiency of the MAC protocol while carrying VoIP traffic.

It is important to note that the WiMAX MAC header and CRC cannot be suppressed or compressed because they are generally not constant and there is no suppression or compression support at the PHY for the upper layers.

4.4.10 QoS Classification

Voice traffic on each network segment of the WiMAX network must be properly classified in order to ensure good call quality. General classification information is described below. Specific classification techniques are hardware specific.

QoS classification must be applied at the WiMAX subscriber to base station link as well as at the WiMAX network to IP network interface. Because the base station may or may not be behind a NAT, it is important to understand the implications of each scenario.

WiMAX QoS classification mainly takes place at layer 3 and layer 4 of the OSI model. Detailed information is provided below.

4.4.10.1 Layer 3

The following are the key layer-3 classifiers used by WiMAX networks [16] in the uplink and downlink directions:

- Destination IP address /Source IP Address– Classification works if the IP address is part of the Voice Platform Network or a carrier which the service provider has added to the classification. Two problems exist with this approach. The first is that any calls where the RTP is handed off to end points are not classified. The second problem is that the carrier list would require updating every time the RTP server for a carrier is changed, added, or removed. However, if the device is behind NAT, the destination IP could always be the Voice Service Provider's SBC and continuous updating of the carrier list would not be required.
- Source IP address /Destination IP Address– Classification only works in this instance if the service provider has knowledge of, through DHCP or static assignment, the IP address of the end-user devices which require prioritization. NAT is a problem with this solution as well.
- DiffServ – When the VoIP devices apply the Differentiated Services (DiffServ) bits of the IP header to VoIP traffic, this would prioritize voice traffic accordingly. Unfortunately, the reverse is not true. Current carriers likely do not tag their packets and even if they did there is no guarantee that the packet markings would "stick" while traversing the public internet.
- Protocol – The classification of UDP protocol does little to distinguish SIP/RTP traffic from any other types of traffic.

4.4.10.2 Layer 4

The following are the key layer-4 classifiers used by WiMAX networks [16] for uplink and downlink directions:

- Source Port /Destination Port – Since the source RTP port range is configurable, classification only works for devices which are not behind NAT.
- Destination Port /Source Port – Classification works toward the Voice Service Provider's SBC, for which UDP ports can be controlled. Carrier connections and end user handoff are both problematic.

4.4.11 WiMAX® QoS Classification Recommendations

It is recommended that a comprehensive approach to QoS classification be taken since no single approach will include all network designs and call scenarios. Recommendations for three areas of traffic classification issues s are provided below in order to cover all call scenarios.

- Prioritizing traffic to the Voice Service Provider's Network – this will classify all devices behind NAT.
- Prioritizing traffic from UDP source/destination ports – this will classify all devices which have a configurable source SIP and RTP range and are not behind NAT.
- Prioritizing traffic to the carrier networks – this will classify the remaining subset of devices which do not have a configurable source port range for SIP and RTP and are not behind a NAT device on the customer premise.
-

4.4.12 WiMAX® Airlink Scheduling, Radio Resource Management and Protocols Implementation Recommendations

In current WiMAX deployments, the airlink transmission is mainly based on TDD (Time Division Duplex) 59 which has a 5 ms frame and is partitioned into a downlink (DL – from the BS towards the MS) subframe and an uplink (UL – from the MS towards the BS) subframe. Figure 4-2presents an overview of a WiMAX TDD frame.

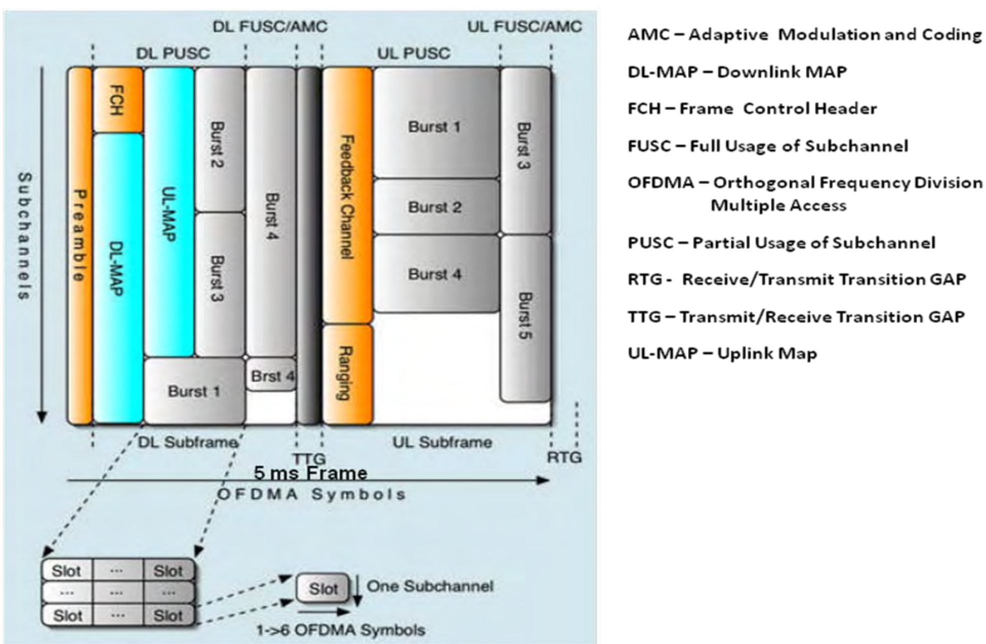


Figure 4-2 : WiMAX OFDMA TDD Airlink Frame Structure [55]

Transmission scheduling over the WiMAX airlink selects the data for transmission in a particular 5 ms frame. Bandwidth allocation is decided by the BS for downlink and uplink corresponding to the service requested by the MS. In addition to whatever other factors the scheduler may deem pertinent, the following items are taken into account for each active service flow in deciding the size of the bandwidth allocation (and hence the time-frequency resources for transmission):

- The QoS service class specified
- The values assigned to the service flow's QoS parameters
- The availability of data for transmission
- The capacity of the granted bandwidth

While the BS can make DL resource allocation decisions based on instantaneous availability of data, on the uplink the BS must rely on bandwidth requests from the MS (except for the UGS and ert-PS scheduling classes). For a given service flow which has been associated with a QoS scheduling class and the specified QoS parameters, the BS scheduler can anticipate the throughput and latency needs of the uplink traffic and provide polls and/or grants at the appropriate times to provide each MS with bandwidth for uplink transmission or opportunities to request additional (or reduce) bandwidth. This ensures that the uplink resources allocated to the MS are tailored to the current traffic but limited by the envelope specified by the QoS parameters.

4.4.12.1 WiMAX® QoS Scheduling Classes Overview

WiMAX technology supports the following QoS scheduling classes: unsolicited grant service (UGS) which can be used for the constant bit rate (CBR) service, real-time polling service (rt-PS) for variable bit rate (VBR) service, extended real-time polling service (ert-PS) for VoIP service with silence suppression, non-real-time polling service (nrt-PS) for non-real-time VBR, and best effort service (BE) for service with no rate or delay requirements. These QoS classes are associated with certain predefined sets of QoS-related service flow parameters, and the airlink scheduler supports the appropriate data handling mechanisms for data transport according to each QoS class. Table 4-2 : QoS Service Classes summarizes the QoS category and the associated QoS service flow parameters.

Table 4-2 : QoS Service Classes [55]

QoS Service Class	E.g. Applications	QoS Parameters
UGS Unsolicited Grant Service	Constant Bit Rate Video or Voice	Maximum Sustained Rate Maximum Latency Jitter Tolerance Traffic Priority Grant Interval (for uplink)
rtPS Real-Time Polling Service	Streaming Audio or Video	Minimum Reserved Rate Maximum Sustained Rate Maximum Latency Traffic Priority Pulling Interval (for uplink)
ertPS Extended Real-Time Polling Service	VoIP with Activity/Silence Detection	Minimum Reserved Rate Maximum Sustained Rate Maximum Latency Jitter Tolerance Traffic Priority Grant Interval (for uplink)
nrtPS Non-Real-Time Polling Service	File Transfer Protocol (FTP)	Minimum Reserved Rate Maximum Sustained Rate Traffic Priority
BE	Data Transfer, Web Browsing,	Maximum Sustained Rate

Best-Effort Service	etc.	Traffic Priority

4.4.12.2 Airlink Scheduling Class Selection Impacts for DL and UL Bandwidth Allocation for VoIP Support

Extended rtPS (ertPS) is a scheduling mechanism tailored to variable rate VoIP traffic with activity/silence suppression. The BS provides unicast grants in an unsolicited manner, thus saving the latency of a bandwidth request. However, ertPS allocations are dynamic and tailored to the instantaneous rate of the VoIP traffic.

By default, the size of uplink allocation corresponds to current value of Maximum Sustained Traffic Rate for the connection. The MS may request changes to the size of the uplink allocation either by using the Extended Piggyback Request field of the Grant Management Sub Header, the Bandwidth Request field of the MAC signaling headers or by sending a codeword over CQICH (Channel Quality Information Channel). The CQICH design provides a fast feedback channel for the MS to report the latest channel conditions to the BS , but also allows the MS to indicate a change in the UL resource allocation in a “piggy back” fashion.

The BS is expected to hold the size of uplink allocations constant until receiving a bandwidth change request from the MS. When the MS sets the Bandwidth Request size to zero, the BS may provide allocations only for a Bandwidth Request Header or the BS may make no allocations at all. In case no unicast Bandwidth Request opportunities are available, the MS may use contention request opportunities to send a bandwidth request for that connection, or the MS may send a CQICH codeword corresponding to a bandwidth request to inform the BS of pending data in the MS buffer. When the BS receives the CQICH codeword, the BS may start allocating uplink grants corresponding to the current Maximum Sustained Traffic Rate value (assuming the BS has sufficient resources). The mandatory QoS parameters are Maximum Sustained Traffic Rate, Minimum Reserved Traffic Rate, Maximum Latency, Request/Transmission Policy and Unsolicited Grant Interval.

Figure 4-3 presents an overview regarding the DL and UL airlink interactions for QoS resource allocation. Note that, the CID (Channel Identifier) as shown below uniquely identifies a given uni-directional service flow within a BS.

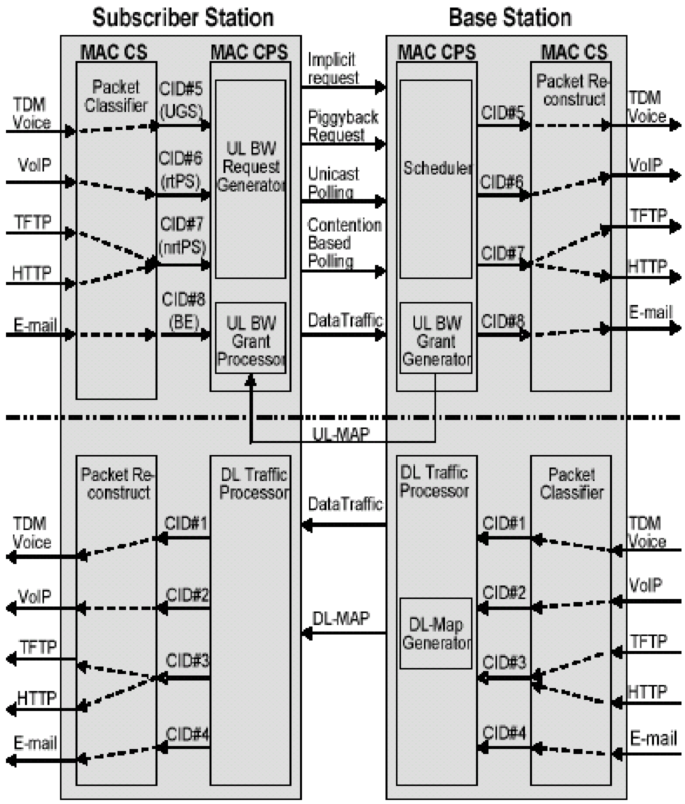


Figure 4-3: WiMAX Airlink Resource Allocation for UL/DL based on the Scheduling Class for a given service flow [55]

4.4.12.3 HARQ

Hybrid Auto Repeat Request (HARQ) is supported by =WiMAX technology. HARQ provides fast response to packet errors and improves cell edge coverage. For delay-sensitive applications such as VoIP, packets received in error are typically dropped and are never retransmitted. Thus, to maintain an acceptable error rate, the modulation and coding level for transmission must adapt to the changing channel conditions. HARQ, however, allows low-latency PHY-level retransmission, and may be employed for VoIP traffic. By carefully limiting the number of allowed HARQ retransmissions, VoIP with HARQ can still meet the One-Way RAN Transfer Delay requirement in Table 5-2 : VoIP Service Performance Parameters.

5 Key Performance Indicators (KPIs)

KPI's are commonly used in business terminology to assess the performance of a given field, in which different metrics are defined to measure the goals set. For VoIP service guidelines, KPI's are defined so that the performance of the voice communication over WiMAX® networks is better understood. The KPI metrics are defined under two main categories: Network KPI's and Application KPI's. While network KPI's can be mostly used to assess the performance of core WiMAX networks, the Application KPI's can be used by service providers to set their business goals [7]. Each of this main KPI's are further analyzed with sub KPI metrics and when applicable target values and the means to measure are presented.

5.1 Network KPIs

5.1.1 Data Rate

A typical voice channel uses 64 kbps at the IP layer based on G.711. Depending on the coding used, the required data rate can change. However, as recommended in section 4.4.4, G.729 can be used and the required data rate would be significantly less than 64 kbps dependent on the sampling rate, e.g. 20 ms or 30 ms etc..

5.1.2 Latency

Delay characteristics of WiMAX systems, as studied in [57], is noteworthy as it can significantly influence the overall performance of the users. To measure the round trip time a small UDP packet can be sent over the WiMAX link and the time it takes for the response for the packet sent is determined as the round trip time. The procedure can be repeated for averaging.

The delay can be introduced via the network itself, CODEC processing, and polling. The recommended one-way E2E (from mouth-to-ear) latency is 160 ms or less for un-noticeable voice degradation. For more explanation of these parameters, refer to section 4.2.1.

5.1.3 Packet Loss

Packet loss occurs when the transmission of some or all packets fails to reach their destination either due to noise or broken network links. As described in section 4.2.2, the recommend packet loss [56] for VoIP is no more than 1 %.

5.1.4 Jitter

Jitter is a variation in packet transit delay caused by queuing, contention and serialization effects on the path through the network [11]. In general, higher levels of jitter are more likely to occur on either slow or heavily congested links. High levels of jitter cause large numbers of packets to be discarded by the jitter buffer in the receiving IP phone or gateway. This may result in severe degradation in call quality or large increases in delay.

Jitter can be measured via the method described in IETF RFC 1889 [20] or the Jitter Envelope measurement provided by VQmon, which is a multi-platform, multi-vendor technology for measuring the quality of Voice over IP services and providing diagnostic data for problem resolution

As explained in section 4.2.3, jitter levels should be less than 50 ms, although < 20 ms is preferred.

5.1.5 VoIP Capacity

Although a specific number on the VoIP users per sector depends on several parameters like bandwidth and their service level agreements (SLAs), supporting at least 200 VoIP subscribers per sector simultaneously, as specified in the WiMAX Forum's IMT-2000 submission to ITU-R on Data and Voice Capacity, [57] can be a starting target. The key to the number of subscribers is not to deteriorate the other KPI's like latency, packet loss, and call drops. It is desirable to develop a KPI based on the existing KPI's, however such studies are still underway. VoIP capacity can be measured by counting the maximum number of simultaneous bi-directional calls in a sector (in term of Erlangs/MHz/cell at 1% blocking as specified in [58]), each of which should meet the requirements like packet loss =< 1%, jitter < 50 ms, etc.

5.1.6 Handoff Success Rate (HSR)

Handoff success rate (HSR) defines the success of WiMAX users being transferred from one base station to another when certain conditions are met. We define HSR to be better than 96 %. Although some simulations method can be developed, more realistically HSR can be measured by analyzing the logs of the network management server and identifying the success rate.

5.1.7 Call Success Rate (CSR)

The Call Success Rate (CSR) is one of the most important (KPIs) used by all mobile operators to measure network quality from a subscriber's perspective. CSR measures the number of calls for which routes are successfully obtained through the network. We recommend CSR to be better than 98%. Although there is no standard measurement possible for this parameter, simulation models and the network management logs can be used to identify this KPI.

5.1.8 Dropped Call Rate (DCR)

The dropped call rate is the percentage of calls that are dropped during the call period which are caused specifically by the operator's network, usually due to the loss of radio link. DCR rates between 2-5% are good indicators of this KPI and it can be measured via the methods of HSR.

5.1.9 Summary – WiMAX® E2E Performance for VoIP Support

The WiMAX network supports VoIP services. Voice quality expectations for VoIP services may vary with the usage scenarios. In general, WiMAX voice quality is required to be equal to or better than the voice quality of current 3G networks in the case of the Full Mobility usage scenario, and is also required to be equal to or better than the voice quality of current VoIP services over DSL and cable for the Fixed and Nomadic usage scenarios.

Call setup latency is particularly important for regular VoIP and for PTT scenarios. WiMAX VoIP services support the call setup and latency requirements listed in Table 5-1 : Call setup and Latency Requirements for VoIP Services.

Table 5-1 : Call setup and Latency Requirements for VoIP Services[2]

VoIP Service Attribute	Target to which WiMAX network conforms
VoIP Call signaling setup from invite to alert	< 1.5 s
Latency from answer to start of voice media stream	< 500 ms

5.1.9.1 Delay, Jitter, Packet Loss

For VoIP, typical application requirements such as bandwidth, delay or latency, jitter and packet loss rate are listed in Table 5-2 : VoIP Service Performance Parameters. The Mobile WiMAX profile for TDD specifies support of at least 200 simultaneous VoIP calls per sector [58], using a channel bandwidth of 10 MHz with a VoIP packet drop rate of less than 1%. The packet drop rate of 1% excludes VoIP packets with more than 250 ms mouth-to-ear latency and packets which are dropped over the air interface. 250 ms of mouth-to-ear latency corresponds to mobile-to-mobile latency and 100 ms out of 250 ms is for air

[2] In the table, all values represent 95%-ile points.

interface latency. The VoIP capacity is based on the assumption that the G.729 8 kbps (45B VoIP packet including 3-byte header compression, 12-byte AES encryption, and 2-byte HARQ CRC overheads) with Voice Activity factor 0.5 (2-state Markov model).

Table 5-2 : VoIP Service Performance Parameters.

Bandwidth	End-To-End One-Way Delay	Packet Delay Variation	Information Loss (PER)	One-Way RAN Transfer Delay
4 – 64 kbps	< 160ms (preferred) < 200 ms (Max)	< 20 ms (preferred) <50 ms (Max)	< 1%	< 25 ms

5.2 Application Specific KPIs

5.2.1 Perceptual Evaluation of Speech Quality (PESQ)

Standardized on ITU-T recommendation P.862 (02/01), PESQ is a set of tests developed for speech quality. The tests assess the voice quality in telephony systems that compromises the voice source, network equipment, and the end user devices. Hence, the PESQ test results can be taken as KPI's for the VoIP service guidelines.

The key tests of PESQ are Sidetone Distortion, 3GPP sending Handset, and 3GPP Receiving Handset, max. Volume [7]. The sidetone distortion test determines 3rd order harmonic distortion at a given set of frequencies (in the case of 3GPP TS 51.010 at 315, 500 and 1000 Hz) with a level of -4.7 dBPa at the MRP. The 3rd order harmonic distortion is required to be less than 10% (-20dB) at all frequencies.

In 3GPP Sending Handset testing, a weighted average of the frequency response is calculated to provide a single number representing perceived loudness of the telephone. The nominal loudness ratings are specified in the network loss plan to ensure that loudness is consistent between different telephones on different connections and that there is adequate head room to accommodate peak signal levels.

Similar to sending Handset testing, in 3GPP Receiving Handset testing, a weighted average of the frequency response is calculated to provide a single number representing perceived loudness of the telephone. The nominal loudness ratings are specified in the network loss plan to ensure that loudness is consistent between different telephones on different connections and that there is adequate head room to accommodate peak signal levels.

5.2.2 R-Score

R-score represents the quality of VoIP calls based on the packet loss and delay of VoIP packets [9]. The loss is more sensitive than delay and, hence, it is key to try to recover as many dropped packets as possible, at the cost of increased delay as long as it is less than the required value. Typical values of R-score of 70 represent a good voice quality.

6 WiMAX Forum® Standardized VoIP Network Solutions Overview

WiMAX Forum® has been working on two separate specifications that support VoIP over WiMAX® networks:

A. IMS-based support for voice and multimedia applications, which is part of the WiMAX Forum Network Architecture Release 1.5 and is based 3GPP Release 7.

B. Non-IMS-based support for voice , which will be completed included in WiMAX Forum Network Architecture Release 2.0.

The following sections present an overview on these two WiMAX standardized VoIP network solutions.

6.1 Non-IMS based VoIP over WiMAX® Networks

6.1.1 Non-IMS based WVS Reference Model

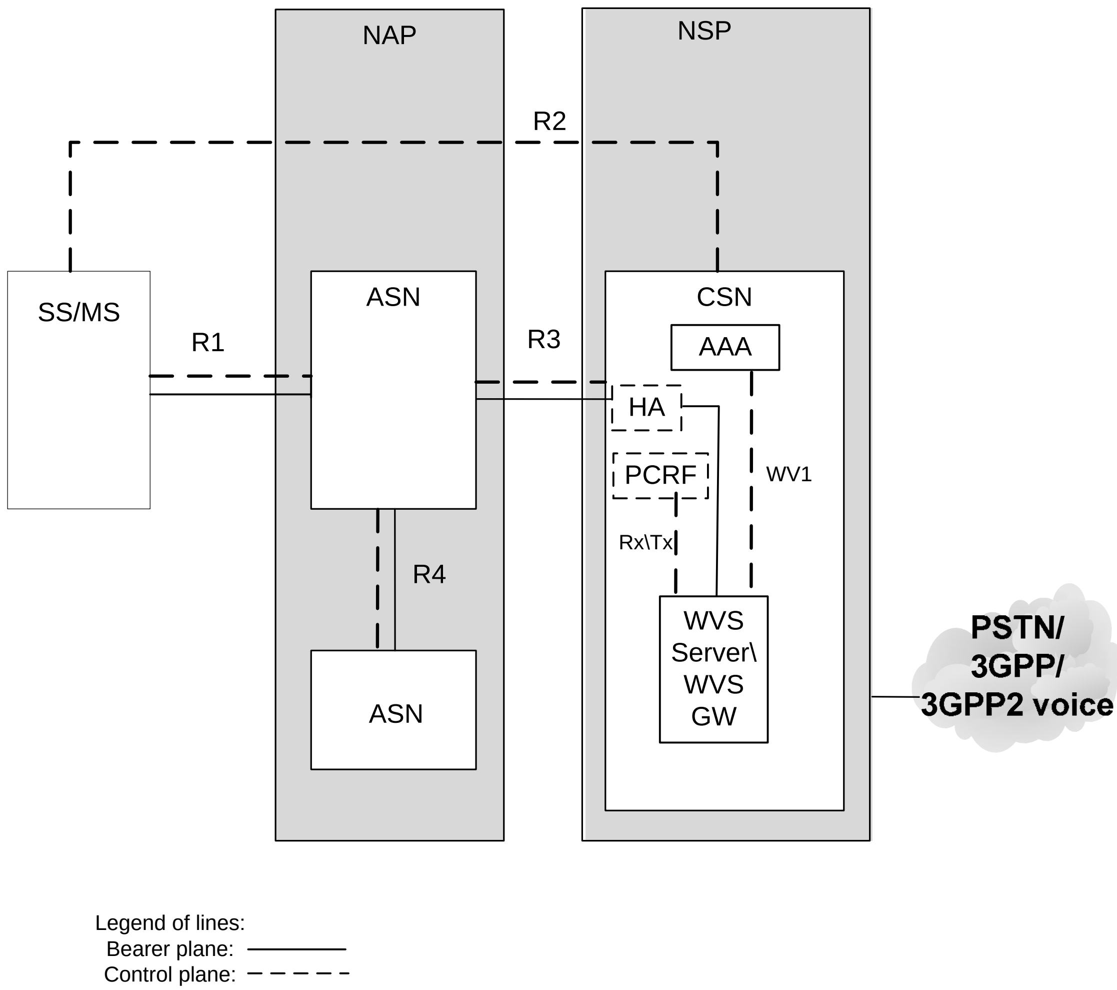


Figure 6-1 : Non-IMS based WiMAX VoIP Service Network Reference Model

WVS Server:

WVS Server includes the functionality of SIP UAC [59], SIP UAS [59], SIP Registrar , Redirect Server , SIP Signaling Transition Function, QoS and policy distribution function (e.g. 3GPP PCC Rx [23]) and AAA Client.

Note: The QoS and Policy could be dynamically or statically configured. AAA Client is only for WVS.

WVS Gateway:

A WVS Gateway includes the transcoding functions and provides bearer connectivity with the PSTN.

The WiMAX CSN supports an interface to the PSTN for signaling and data transfer with or without any transcoding operation. This interface is out of scope of WiMAX Forum specifications.

WiMAX PCC [60] is used for enabling QoS support

6.1.2 Non-IMS based WVS Reference Points for Control Plane

Reference Point WV1

It is the interface between WVS Server and AAA Server.

It is RADIUS or DIAMETER based. RADIUS is mandatory and DIAMETER is optional.

Reference Point R2

It is the interface between MS and WVS Server and is SIP-based.

6.2 Non-IMS Based WiMAX® VoIP Security Support

6.2.1 User Identity for Security

Each WVS user will be provided with one Inner User Identity. The inner identity is assigned by the H-NSP and is used, for example, for Registration, Authorization, Administration, and Accounting purposes. This identity shall take the form of a Network Access Identifier (NAI) as defined in RFC 2486 [61] Security over the Air

Since all SIP signalling and voice packets are data plane payload packets over the air in IEEE Std 802.16-2009 [56], the security mechanism defined in [56] applies. Hence, both of the WVS control plane and data plane can be protected by encryption.

6.2.2 WVS Authentication and Authorization Method

HTTP-Digest as defined in [61][62] is used by non-IMS WVS for authentication and authorization.

6.3 IMS-based VoIP over WiMAX® Networks

IMS is an open standardized multi-media architecture for mobile and fixed IP services originally defined by 3GPP. This section explains on how VoIP service is supported by IMS Release 7 [63][64][65][66]over a WiMAX network. Figure 6-2 shows non-roaming reference architecture of WiMAX network with IMS and PCC.

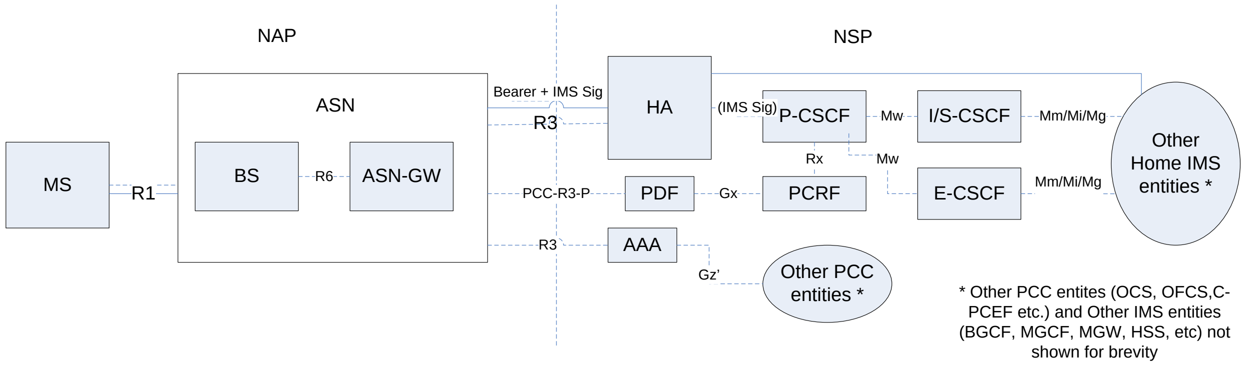


Figure 6-2 : IMS-based WiMAX® VoIP Service Network Reference Model

P-CSCF

The Proxy-CSCF (P-CSCF) is the first entry point within an IMS subsystem. It forwards the VoIP SIP request/response messages including registration messages and call set up/release messages to an I-CSCF or S-CSCF or forward them on towards the destination network. It is responsible to detect and handle an emergency session establishment request. The P-CSCF is responsible to generate the CDR, and maintain a security association between itself and the MS.

I-CSCF

The Interrogating-CSCF (I-CSCF) is the first entry point within an operator's network for all connections destined to a user of that network operator. During IMS registration, it assigns an S-CSCF for a MS and forwards the SIP message to the S-CSCF. It performs the HSS location Query.

S-CSCF

The Serving-CSCF (S-CSCF) performs the session control services for the MS. It controls and maintains a VoIP session state as needed by the network operator for support of the service.

When a caller initiates a call, the MS sends a SIP call request message to the P-CSCF. The S-CSCF checks and ensures that the caller is subscribed to the IMS service. If the caller does have a subscription, and if the caller is at a different network, the S-CSCF is responsible to find the entry point of the destination network and forward the message to that entry point. It is responsible to forward the SIP request or response to a BGCF for call routing to the PSTN or CS Domain.

The S-CSCF checks and ensures that the caller is subscribed to the IMS service. If the caller does have a subscription, it forwards the message to the P-CSCF in the caller’s network. It Forwards the SIP message to a BGCF for call routing to the PSTN or to the CS domain.

E-CSCF

The Emergency-CSCF (E-CSCF) is responsible for routing emergency call requests to the nearest PSAP based on the caller’s location information,

PCRF

The PCRF encompasses policy control decision and flow based charging control functionalities. The PCRF performs session binding (i.e., the association of the VoIP session information and applicable PCC rules to an IP-CAN session) and PCC rule authorization.

PDF

The PDF hides the distributed nature and mobility of WiMAX enforcement points from the PCRF. The PDF is connected to the Anchor SFA in the ASN via PCC-R3-P interface and supports SFA relocation by decoupling PCC-R3-P sessions from Gx/Ty.

The PDF is the distribution point for the PCC rules between the ASN and the CSN; it proxies the Gx (Ty) messages from the PCRF over the PCC-R3-P interface between the PCRF and the A-PCEF. The interface between the PDF and C-PCEF is out of scope for the present release of [60].

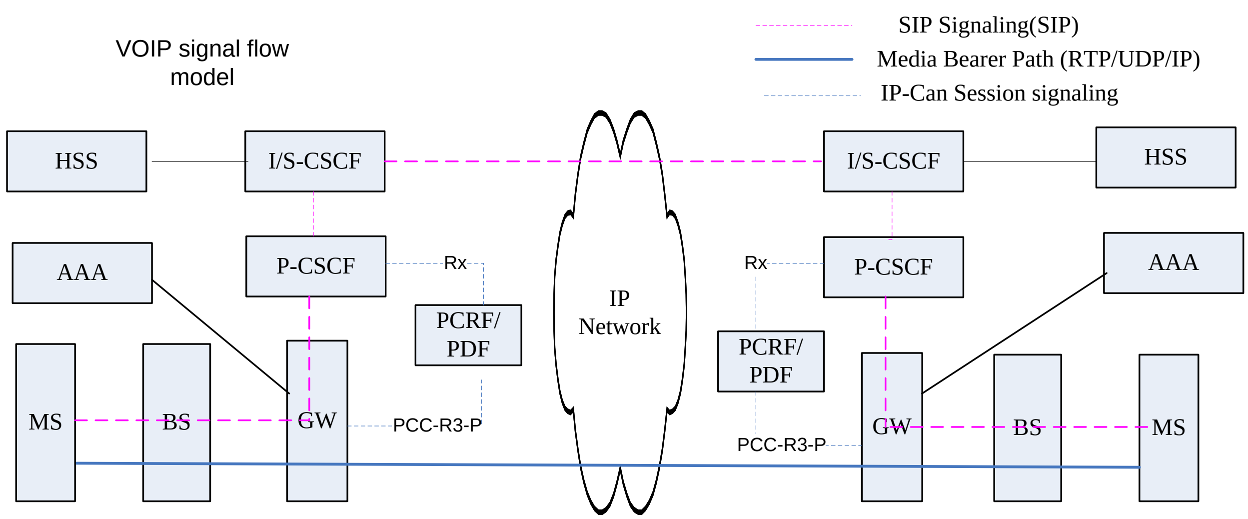


Figure 6-3 : IMS Integration with Policy and Charging Control (PCC) framework Network Reference Model – Depicting a mobile to mobile call

After device and user authentication, the MS enters the WiMAX network. The MS performs P-CSCF discovery procedure and obtain P-CSCF address, then the MS performs IMS registration if it is not registered. The MS makes a VoIP call by sending a SIP signaling message which includes the media information and QoS information. When the P-CSCF received the message, it checks if the call is an emergency call, if it is, then it will forward the message to the Emergency-CSCF (details in next section). If the call is not an emergency call, it forwards the message to the MS's S-CSCF. The S-CSCF finds the caller's network entry point and forwards the message to that entry point. The P-CSCF also passes the media QoS information to the PCRF to trigger IP-CAN session modification at the caller side. The PCRF is responsible to generate the rules and charging policy and delivers to the GW so that GW can modify or build a bearer path for the VoIP call. The caller's network entry point, I-CSCF, receives the SIP message from the caller's S-CSCF, the I-CSCF forward the message to the caller's S-CSCF in its network. The S-CSCF checks and ensures that the caller subscribed to the IMS service, and forwards the message to the P-CSCF. The P-CSCF in the caller's network passes the media QoS information to the caller's PCRF to trigger IP-CAN session modification, the PCRF is responsible to generate the rules and charging policy and deliver to the caller's access network GW so that the GW can modify or build a bearer path from the caller side to support the VoIP call.

Emergency VoIP call supported with IMS

The WiMAX network with IMS enables emergency service call to be delivered to the nearest emergency network. It supports the emergency service caller is a regular IMS user using a valid device; it also supports the emergency service call made by a MS which is not allowed to enter the WIMAX network for other purpose other than emergency. The following diagram shows an IMS emergency service network reference model for a non-roaming user.

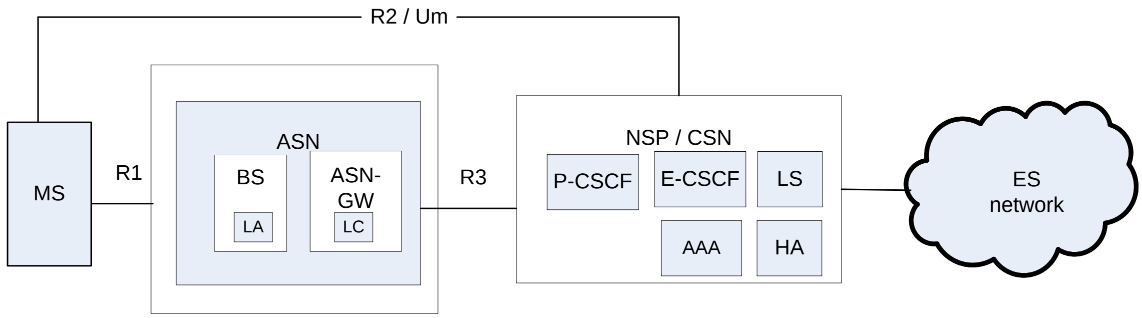


Figure 6-4 : IMS-Based Emergency Service Support for non-roaming VoIP user

Note: Almost all access networks supporting Emergency Services today are PSTN based networks.

The Location Agent (LA) function in the BS, the Location Client (LC) function in the ASN-GW, along with the Location Server (LS) in the CSN are location retrieval functions that collaborate to determine the geo-location coordinates or civic address of the distressed MS. When the MS initiates the emergency call by sending a SIP INVITE message with a ES URN, the P-CSCF determines that it is an emergency call and forwards the SIP message to the E-CSCF, meanwhile the P-CSCF also triggers the PCRF to create a high priority service flow and a data path for the call. The E-CSCF may enquire the location retrieval functions to find a routing number to an appropriate ES network for the WIMAX MS device originating the ES call. The emergency call is set up between MS and the ES network nearest to the MS. Note: the ES network may enquire the location retrieval functions to get updated location information of the MS.

More additional design considerations for VoIP support for ES are described in latter section 10.

6.4 Policy control

The WiMAX Policy and Charging Control (PCC) are largely based on the 3GPP Release 7 PCC framework. Figure 6-5 below presents the 3GPP PCC Release 7 architecture.

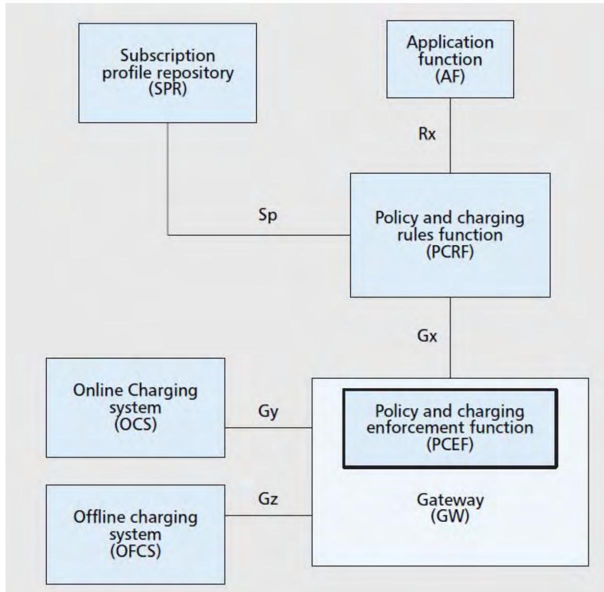


Figure 6-5: 3GPP PCC Release 7 Architecture

Note that WiMAX bearer transport is "connection oriented"[3] and the connection is referred as the concept of "service flow" which is applicable only to the WiMAX Access Serving Network (ASN).

A WiMAX service flow is uni-directional and is uniquely identified by its corresponding classifiers at each endpoint of the WiMAX access network – i.e. at the MS to classify the packets for the uplink and at the ASN to classify traffic for the downlink.

Fundamentally, the WiMAX ASN is agnostic to the service application that utilizes a given service flow. The QoS enforcement for a given service flow is based on the network policy, as defined by PCC, that is associated with the classifiers which uniquely identifies the service flow. However, WiMAX specifications allow the optional indication to the ASN for the media flow type that applies to the given service flow in case that the local implementation requires additional information to drive the operational policy (e.g. power saving support).

At the WiMAX Core Serving Network (CSN) which interconnects to the ASN over the R3 reference point, the bearer transport is carried over the service data flows and packet data flows (see Figure 6-6: Mapping WiMAX Flows to PCC Flows below) that can be bi-directional and are designed to bundle the WiMAX service flows at the ASN for a given application.

The following figure describes the mappings of the application service data flows to the WiMAX CSN service data flows to WiMAX ASN service flows according to the WiMAX PCC framework.

[3] In telecommunication, *connectionless* describes communication sent from one end point to another without prior arrangement. The device at one end of the communication transmits data to the other, without first ensuring that the recipient is available and ready to receive the data. The device sending a message simply sends it addressed to the intended recipient. The Internet Protocol (IP) and User Datagram Protocol (UDP) are connectionless protocols. Connectionless protocols are usually described as stateless because the endpoints have no protocol-defined way to remember where they are in a "conversation" of message exchanges. The alternative to the connectionless approach uses connection-oriented protocols, which are sometimes described as stateful because they can keep track of a conversation.

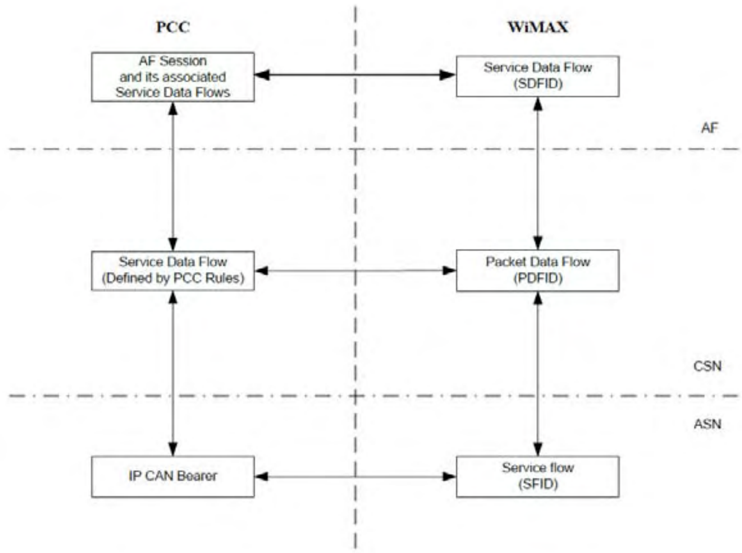


Figure 6-6: Mapping WiMAX Flows to PCC Flows

In general, WiMAX service flows can be categorized as either pre-provisioned or dynamic. Pre-provisioned service flows are established at the time of initial network entry as authorized by the home AAA server of the subscriber. They are comparable to "default" flows in the 3GPP-defined QoS architecture. In contrast, dynamic service flows are established, modified or terminated as the result of a real-time trigger from the application layer. They are comparable to "dedicated" flows in the 3GPP world.

The WiMAX Release 1.0 network specification ([13][14]) supports only supports pre-provisioned service flows. As the QoS assignment and the classifiers for the pre-provisioned service flow cannot be changed dynamically, pre-provisioned service flows may only be used to support fixed WiMAX deployment for VoIP in the case when the SIP Proxy server is statically provisioned for the given WiMAX device, and the codec is also statically configured.

The WiMAX Release 1.5 network specification introduces support of dynamic service flows as well as the Policy Charging and Control (PCC) framework. Dynamic service flows can make use of the policy exchange framework that is enabled by the WiMAX Policy and Charging Control (PCC) architecture as shown in following Figure 5-3 to support all types of WiMAX deployments (i.e. fixed, nomadic and mobile) for VoIP.

WiMAX PCC was introduced in Release 1.5, it can support both pre-provisioned and dynamic service flows.

6.4.1 Benefits of WiMAX® Release 1.5 PCC over 3GPP Release 7 PCC

WiMAX Release 1.5 PCC framework is largely based on the 3GPP Release 7 PCC because the 3GPP Release 8 PCC was still work in progress at the time when WiMAX Release 1.5 was developed. However, WiMAX Release 1.5 introduced its own enhancements to the Policy Control and Charging framework for the following main reasons and benefits:

- Because of the lack of inter-RAT mobility and roaming support in 3GPP PCC Release 7 – which assumes a single fixed PCEF at the core network to support the lifetime of an IP session for a given device, the WiMAX Forum introduced the Policy Distribution Function (PDF) along the path between the PCRF and the PCEF to minimize the deviation from the original 3GPP PCC framework. The main role of the PDF is to hide the specifics of the WiMAX access network design (e.g. such as the QoS policy mapping and the inter-RAT mobility anchoring) from the standard PCRF implementation.

- Through this logical functional entity, WiMAX networks can maintain compatibility with the 3GPP PCC design. The PDF also enables its own flavor of R3-PCC interface to allow backward compatibility with the WiMAX Release 1.0 R3 QoS framework. The WiMAX R3-PCC interface enables support for a DIAMETER-based PCC interface and a RADIUS-based PCC-like interface. This design decision is critical for many WiMAX operators around the world which are still in favor of deploying RADIUS rather than DIAMETER for their policy control and charging support.
- Another major benefit of the WiMAX PCC framework is to enable the support of the charging and accounting support not only at the access network via the support of the A-PCEF function, but also at the home core network of the subscriber via the optional support of the C-PCEF function at the HA. This is an important network architecture option to enable the ASN-sharing by different Network Service Providers (NSPs) to be able to enforce their independent policy control, charging and accounting functions for their own subscribers attaching to the a physical radio access network which may be operated by a different Network Access Service Provider (NAP).
- For QoS activation, WiMAX networks support both the PUSH and PULL models instead of the PULL ONLY model preferred in 3GPP Release 7. In the PUSH model, the PCRF triggers enforcement of QoS requirements at the access network when service information is received from the AF (e.g. SIP server/proxy). Hence, the MS does not need to perform QoS reservation, which is necessary for the PULL model. This allows for optimization of QoS settings by the network and simplifies the application client development on the terminals. As a consequence, support for network-initiated QoS enforcement is mandatory whereas terminal-initiated QoS enforcement is optional.
- Finally, by preserving compatibility with the 3GPP Release 7 PCC interfaces, packet core network and service policy convergence becomes feasible for inter-RAT mobility between WiMAX and non-WiMAX access. More on the inter-RAT handover support for VoIP will be discussed in the latter section.

The following figure presents a simplified Mobile WiMAX PCC architecture as described above.

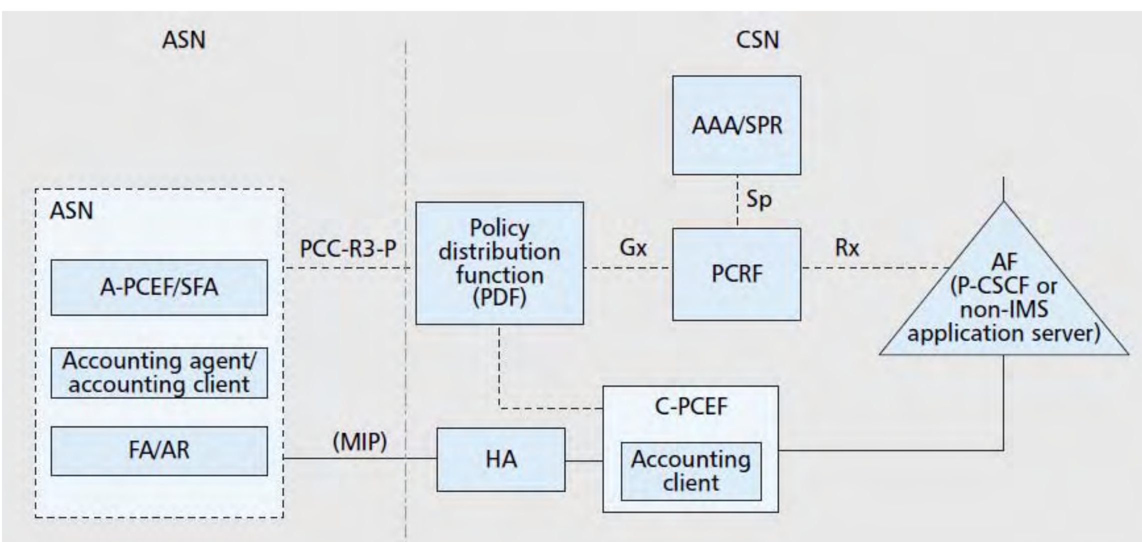


Figure 6-7: Simplified Mobile WiMAX Release 1.5 PCC Architecture

The following summarizes how WiMAX PCC functional components as supported by the WiMAX Release 1.5 PCC architecture [60] shown above support the policy and charging control for both IMS-based and non-IMS-based VoIP service.

The SPR function provides QoS-specific profile information to the PCRF for subscription-based policy decisions. The SPR may be collocated with the AAA server. The AAA server is responsible for access network's authentication, security and accounting for subscribers. By allowing these two functional entities to be logically separated but physically co-located, this approach provides the flexibility to continue the use of SPR for IMS-based VoIP support, while the AAA server for the existing WiMAX network may be upgraded to incorporate the SPR function for the non-IMS based VoIP support.

The PDF is a WiMAX-specific logical entity on the path between the PCRF and PCEF. The PDF was introduced to support the relocation of the policy enforcement point in the access network (i.e. A-PCEF) and distribute policies to the C-PCEF. Furthermore, the PDF is the entity in charge of mapping the 3GPP Release 7 Gx interface to the WiMAX-specific PCC-R3-P interface when interworking with a 3GPP PCRF. The PDF can be a standalone network entity or integrated with PCRF. This design architecture allows the VoIP service interworking during the inter-RAT handover while maintaining the compatibility of the policy and charging control support.

The A-PCEF is the mandatory policy and charging enforcement point and is located at the ASN-GW. The A-PCEF encompasses all the function of the PCEF as defined in 3GPP PCC [21]. The PCEF utilizes PCC rules to classify the traffic carried in each service data flow. The PCC Rules can be either pre-defined, pre-provisioned in the PCEF or dynamically assigned. In addition, A-PCEF is responsible for translating the policy and charging rules received from the PCRF/PDF to the WiMAX-specific QoS and charging attributes – i.e. PCC QoS identifier values into the IEEE Std 802.16-2009 QoS parameters settings and performing accounting according to the received charging rules.

The C-PCEF is an optional enforcement point located in the CSN. The C-PCEF allows CSN-based accounting and gating control in the core, but does not perform any QoS enforcement.

The PCC interface on R3 (PCC-R3-P) is defined based on the Gx interface as specified in 3GPP Release 7. The differences from the Gx are limited to extension of the QoS parameters allowing full support of WiMAX-specific UGS and eRT-PS service types, which are specifically optimized for real-time service flows such as to support VoIP, and the transport of fixed-size data with periodic arrival similar to T1/E1 and circuit-switched connections.

Additional modifications to Gx procedures were introduced in PCC-R3-P to add support for PCC session re-anchoring in the case of A-PCEF relocation. While Gx is based on DIAMETER only, PCC-R3-P allows support of RADIUS as an option in addition to identifying DIAMETER as mandatory to support.

7 WiMAX Forum® Standardized Inter-RAT Handover Support for VoIP

Interworking can be categorized into two kinds – a "nomadic" approach and a "full mobility" approach. In the nomadic approach, "session continuity" between different access technologies is not required. That is, the data and VoIP sessions that exist in the network of one access technology will not be carried over to the other technology, when the user switches between the two. Hence, IP and VoIP sessions that exist in the first network are terminated in the source system before the user enters the target system. However, the market demand for wireless broadband with full mobility including VoIP session continuity is increasing with the emergence of Mobile Internet Devices. Hence, it becomes essential to support interworking with "full mobility" and seamless voice session continuity as the user crosses boundaries of WiMAX® coverage areas. In the "full mobility" seamless handover approach, the voice users must be able to maintain their IP session and corresponding services with service continuity, without experiencing any significant degradation in the voice quality.

The main requirement for achieving seamless integration of WiMAX networks and 3GPP/3GPP2/Wi-Fi access networks is to preserve the VoIP quality provided by the VoIP platforms or the IMS system by minimizing the handover interruption and preserve the quality-of-service (QoS) of the voice session as the mobile terminal moves between mobile WiMAX and the cellular or Wi-Fi access technologies. The objective is to make the transition of the voice call from one access network to another as transparent as possible to the user, i.e., to offer a *seamless mobility* experience with no discernable interruption. With seamless mobility, users can exploit the availability of several access technologies to best meet their charging and VoIP requirements, e.g. by automatically selecting the “most appropriate” access network type based on some pre-configured preference settings and operator policies. At the same time, operators may exploit the VoIP seamless mobility in order to offer compelling value-added ubiquitous services as well as improve their network capacity and availability of services.

Seamless mobility can be achieved by enabling mobile terminals to conduct seamless handovers across mobile WiMAX and 3GPP/3GPP2/Wi-Fi access networks, i.e., seamlessly transfer and continue their ongoing sessions from one access network to another. A seamless handover is typically characterized by two performance requirements: (1) the handover latency should be no more than a few hundreds of milliseconds and (2) the QoS for the voice session provided by the source and target access systems should be nearly identical. These two performance requirements are not trivial to satisfy when mobile WiMAX and 3GPP/3GPP2/Wi-Fi access networks are combined in a common or possibly single architecture. In order to offer seamless handover, several issues need to be addressed. One of them is how fast the service data flow (i.e. the stream of data or voice packets associated with an ongoing service) can be switched from the old path through the old access network to the new path through the new access network. Usage of common or similar link-layer access network procedures (e.g. authentication, security and mobility procedures) in both the source and the target access network can considerably expedite the handovers. One of the challenges associated with the integration of mobile WiMAX and 3GPP/3GPP2/Wi-Fi access networks arises from their differences in terms of AAA procedures, QoS mechanisms and mobility protocols. To address these challenges, the WiMAX Forum completed four interworking specifications ([37][38][39] and [40]), where each of the specifications requires a common anchoring point at the HA/LMA in order to enable minimal interruption when the IP and the VoIP sessions are handed over from the source technology to the target.

The inter-RAT handover solution is required to include support for both single radio and dual radio devices. Single radio devices include support for a single transmitter while dual radio devices include support for dual radio transmitters. A dual radio inter-RAT handover solution may be supported with minimal impacts to the access network based on a ‘loosely coupled’ solution. Support for single radio devices requires a ‘tightly coupled’ solution which requires some changes to the WiMAX and non-WiMAX network to minimize handover latency delays. This typically includes support for a Signaling Forwarding Function (SFF) network entity that enables layer 3 tunneling of target network air interface signaling over the serving access network.

The WiMAX Forum®, 3GPP, and 3GPP2 have developed specifications to provide support for inter-RAT handover to/from WiMAX networks. These include [37][38][39][40][41][42][43]and [44]. The WiMAX Forum network specifications also include both single radio [38] and dual radio support for inter-RAT handover between WiMAX and Wi-Fi networks.

One important consideration for inter-RAT handover support for VoIP is the proper mapping of QoS attributes between the serving/target WiMAX and non-WiMAX networks, more specifically between WiMAX and non-WiMAX networks after the handover. QoS mapping tables for matching the VoIP service flows at the disparate access networks are assumed to be an implementation details that are not specified in the standards.

7.1 3GPP/WiMAX® Handoff

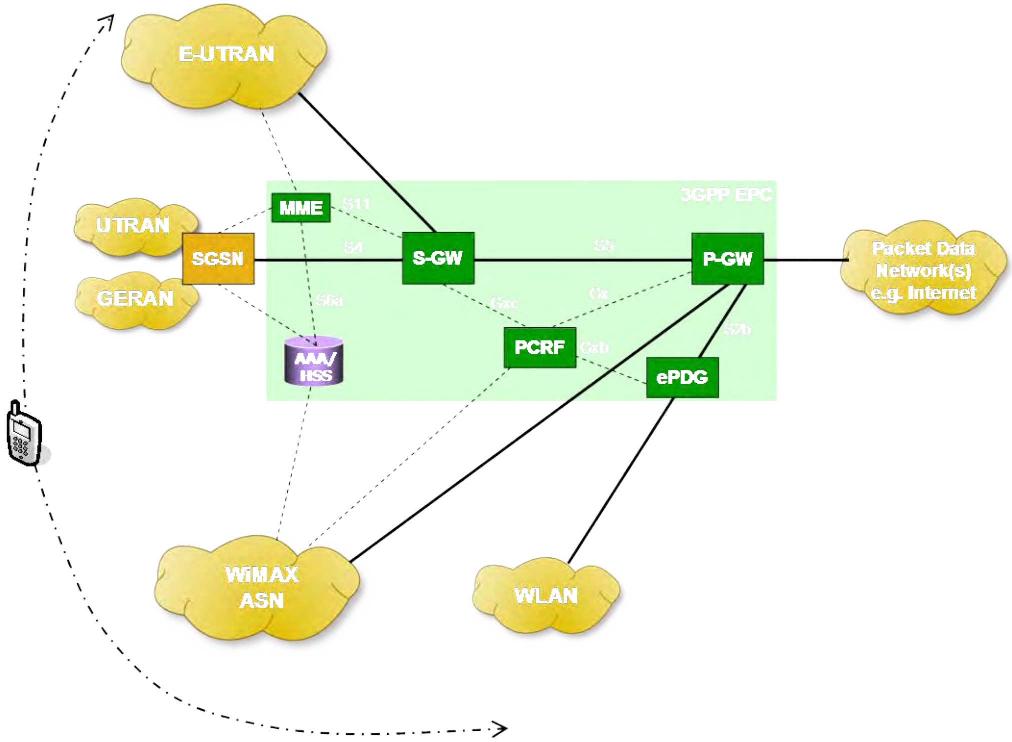


Figure 7-1 : Evolved 3GPP network architecture

As shown in Figure 7.1, in this evolved 3GPP architecture where number of diverse access IP networks, such as WLAN, WiMAX, GERAN, UTRAN and E-UTRAN, are connected to a common core network (the Evolved Packet Core – EPC) through different interfaces. All 3GPP specific access technologies are connected through the Serving Gateway (S-GW), while all non-3GPP specific access technologies are typically connected through the Packet Data Network Gateway (P-GW) or the Evolved Packet Data Gateway (ePDG), which provides extra security functionality for un-trusted access technologies (such as legacy WLANs with no strong built-in security features). The Serving Gateway (S-GW) acts as the mobility anchor for mobility within the 3GPP-specific access technology and also relays traffic between the legacy Serving GPRS Support Node (SGSN) and the PDN Gateway (P-GW). In case of E-UTRAN, the S-GW is directly connected through the S1 interface, while SGSN is the intermediate node when GERAN/UTRAN is used. It is important to mention that a Mobility Management Entity (MME) is also incorporated in the architecture for handling control functions such as authentication, security, and mobility in idle mode. [38]and [40] specifications describe how IP, including VoIP sessions, are handed over using either single radio or dual radio techniques respectively. In a dual radio operation, while connected to the serving technology, one of the dual radio transmitters can establish an IP session on the target technology (e.g. 3GPP), establish the required VoIP service flow and hand over the VoIP session from WiMAX with a minimum interruption. In a single radio operation, while connected to the serving technology, the terminal can pre-register an IP session and set up a VoIP service flow on the target technology and only then handover the VoIP call over. The interruption time of a single radio is higher than a dual radio but both can support a reasonable handover performance with minimum voice quality degradation.

7.2 3GPP2/WiMAX® Handoff

A network architecture that supports full mobility including session VoIP service continuity across WiMAX and EVDO networks is shown in Figure 7-2. The figure shows WiMAX and EVDO access networks sharing the same IP core network, which is directly connected to a common core and through it to VoIP platforms or IMS systems. This is a loosely coupled model because the WiMAX and EVDO networks have separate and independent data paths to the core network. Each network follows its unique network entry procedures, authentication methods, intra-technology mobility, paging, etc. An end user can use the same VoIP platforms and services in either one of the two access networks, since the two access networks provide access to the same VoIP platform or IMS system through the common IP core network. The data paths are separated for WiMAX and EVDO access networks. This architecture supports full IP and VoIP mobility, either for a single radio or dual radio terminal, across the two access networks during inter-technology handoffs by maintaining a MIP tunnel between the HA in the core network and the FA in the access network.

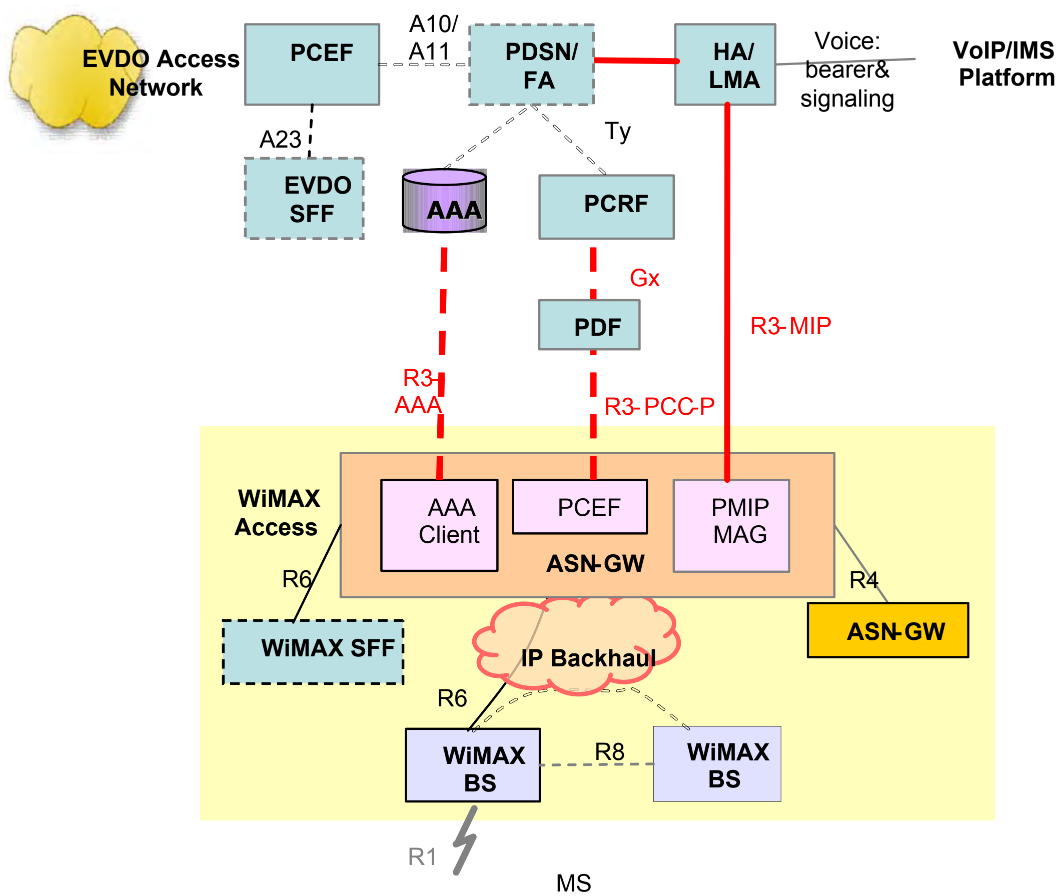


Figure 7-2: WiMAX-3GPP2 Interworking Network architecture

7.3 Wi-Fi/WiMAX® Handoff

An architecture that supports full mobility with VoIP service continuity across WiMAX and Wi-Fi networks is shown in Figure 7-3. The figure shows WiMAX and Wi-Fi access networks sharing the same IP core network, which is directly connected to the VoIP platforms or IMS systems. The Wi-Fi Interworking

Function (WIF) in the diagram includes the following functions; AAA proxy, PMIP client, DHCP proxy, accounting client, FA (or MAG) supporting IP tunnels to the HA (or LMA) and packet filtering.

Like the 3GPP2 case, this is also a loosely coupled architecture since the WiMAX and Wi-Fi networks have separate and independent data paths to a common core network. Since the two access networks have access to the same VoIP platform or IMS system through the common IP core network, an end user can use the same VoIP platforms and voice services when connecting through either one of the two access networks. This architecture supports full IP and VoIP mobility, either for a single radio or dual radio terminal, across the two access networks during inter-technology handoffs by maintaining a MIP tunnel between the HA in the core network and the FA in the WiMAX access network or the WIF in the Wi-Fi network.

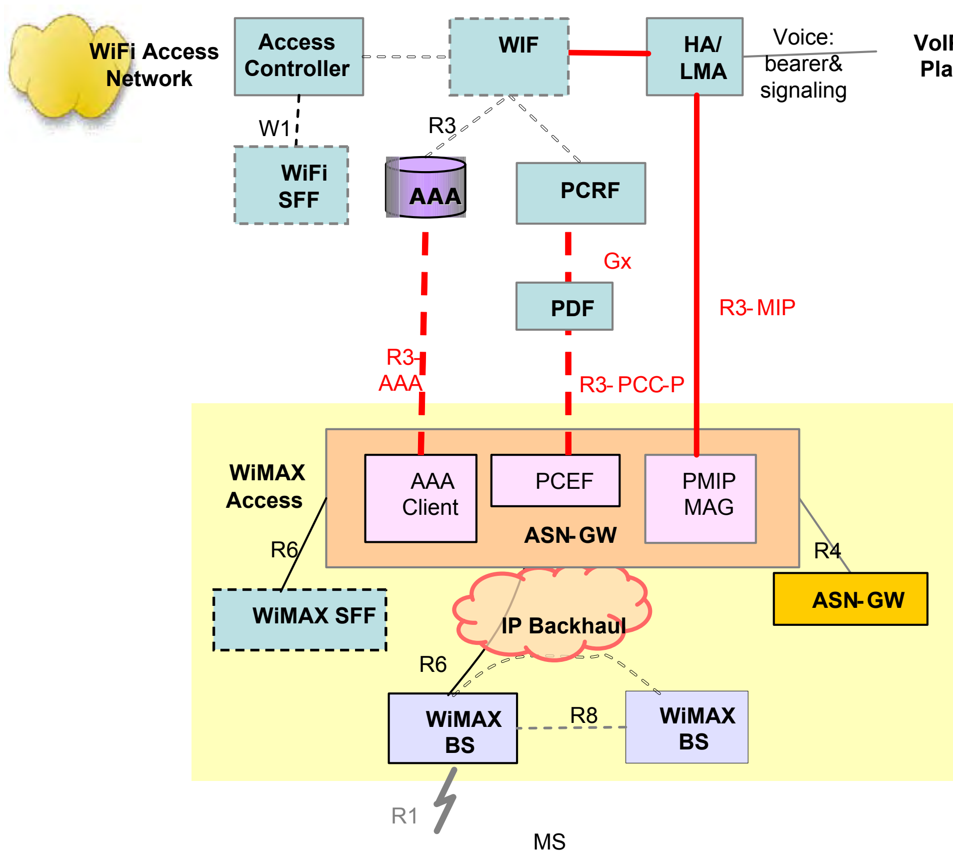


Figure 7-3 : WiMAX®-Wi-Fi Interworking Network architecture

In this interworking architecture, the concept of MIP tunnels is extended to enable interworking during handoff across the WiMAX and Wi-Fi access networks. For example, whenever an MS enters a WiMAX (or Wi-Fi) from the Wi-Fi (or WiMAX) networks, the common HA preserves the address of the MS and creates a new MIP tunnel with the FA in the ASN-GW (or FA in the WIF). Any incoming voice packets to the MS are directed by the HA through the MIP tunnel with the active FA in the respected network. The existing IP sessions (and tunnels) between the device and the core are active trough the active FA and the end result is that there is a VoIP session continuity to the end user during the inter-technology handoffs.

Although not standardized, the Figure above shows how common PCC can be applied to the Wi-Fi network with the additional requirements of adding policy enforcement functionality in the WIF. Without such common policy framework, the handover between the two technologies can only support handing over a voice service flow in WiMAX to a best effort (BE) service flow at the Wi-Fi network. Since the PCC framework in Wi-Fi is not commonly deployed, we believe that initially only BE service can be provided for VoIP users on the Wi-Fi networks. Therefore, in order to maintain acceptable quality voice on the Wi-Fi network, the serving Access Points must be lightly loaded.

7.4 QoS and Policy Control

An important issue for providing seamless VoIP mobility is to maintain a specific level of QoS consistently across the WiMAX and the 3GPP/2 radio access networks. This involves several considerations such as the QoS mappings and semantics on the two access networks as well as appropriate resource allocations. In addition to this, for equivalent VoIP user experience on both networks classification rules must be kept common across both access networks by coordinating the appropriate QoS rules. QoS consistency between WiMAX and 3GPP/3GPP2 access networks can be provided via the PCC infrastructure as defined in the previous chapters. Table 7-1 is an example of QoS parameters mapping between IEEE Std 802.16 and 3GPP2 as captured in [73].

Table 7-1 : Example of QoS parameters mapping between IEEE Std 802.16 and 3GPP2

QoS Attribute	802.16	3GPP2
Throughput	Maximum Sustained Traffic Rate	Peak rate
Packet Loss Rate		Max_IP_Packet_Loss_Rate
Packet Error Rate	Packet Error Rate	
Max Packet Transfer Delay	Maximum Latency	Max_Latency
Packet Delay Jitter	Tolerated Jitter	Delay_Var_Sensitive

8 Device Management and Auto-Configuration and Provisioning

One significant difference between today PSTN network and the 4th generation mobile VoIP network is that, the intelligent mobile VoIP gateway or IP terminal now reside on the customer premises. The mobile VoIP devices need more configuration than a legacy POTS phone. Therefore, the remote auto-configuration, auto-service provisioning and maintenance support for the WiMAX device to support VoIP becomes important requirements to enable scalable deployment and to reduce operational cost associated with service provisioning and system upgrade.

The key device management features for VoIP support are:

- Over-the-air (OTA), remote management and software updates to subscriber devices
- Policy-based auto-configuration, monitoring and debugging of devices
- Fault management and status queries from the device
- Enforcement of varying service tiers and service priorities

- Administration of subscription models such as pre-pay or wholesale arrangements

The WiMAX Forum has standardized the OTA device management specification for WiMAX devices which based on two industry standards:

- OMA-DM – Open Mobile Alliance Device Management [14]
- TR-069 – Broadband Forum (previously DSL Forum) Technical Report 069 [15]

The following are the key generic steps for activating a WiMAX device to attach the WiMAX network:

1. Upon powering the WiMAX device the first time, the device will perform the scanning across all available RF channels to determine which WiMAX service providers are available.
2. Through the support of the WiMAX feature, Network Discovery and Selection (ND&S) (see Annex – A for more details), a list of available WiMAX service providers will be announced and the end-user has the opportunity to select the provider of choice.
3. The connection is immediately re-directed to the operator's service provisioning landing page where key vitals of the end-user are collected including profile information, billing details and subscription preferences.
4. The operator will request the end-user to enter their unique device ID printed conveniently within the package and on the device. The device ID is typically a combination of the MAC address and the serial number of the device.
5. The operator will validate the information and upon election to grant service, the system will then provision the user device onto the AAA and maybe also to other related system elements that the user device is entitled for the service.
6. The end-user is authenticated, security keys are exchanged and the WiMAX broadband connection will then be enabled.

Once the WiMAX device is attached to the WiMAX network, the VoIP service activation and provisioning may then be triggered automatically or by the user (e.g. over the Web browser).

The OMA and Broadband Forums have standardized their respective specifications to support VoIP service provisioning which both are compatible to WiMAX VoIP standard solutions without any additional standardization requirement at WiMAX Forum. For example, Broadband Forum has published TR-104 [TR-104] based on the following VoIP Provisioning Data Model (as shown in Figure 8-1 : TR-104 VoIP Provisioning Data Model) to support the various VoIP service provisioning scenarios.

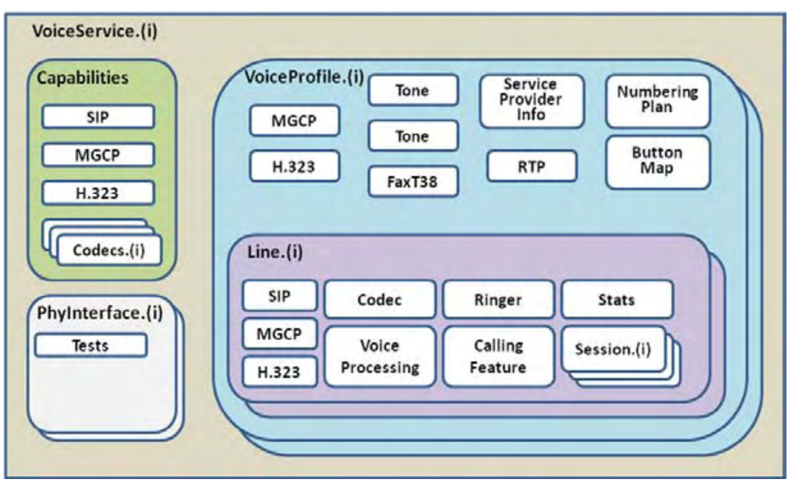


Figure 8-1 : TR-104 VoIP Provisioning Data Model

WiMAX VoIP solution is SIP-based; hence, the WiMAX VoIP service provisioning will refer to the TR-104 [71] SIP-based parameters, if TR-104 is used.

In summary, the standard procedures to support the device management for the VoIP auto-configuration and service provisioning are Service Provider's VoIP deployment decision. The two well-known industry standards, OMA-DM and TR-104, complement the WiMAX VoIP solutions, as described in this white paper.

9 Lawful Intercept (LI)

Lawful Intercept (LI) allows Law Enforcement Agencies (LEAs) to intercept voice and data traffic generated by or directed to a subscriber, regardless of the access technology used. Typically, the LEA sends a request to the network operator to monitor both session data (information about the connection) and session traffic (the actual IP stream) that relate to a specific subscriber.

While the goals of LI are the same across countries, requirements and interface specifications vary. Network Operators need to adopt a solution that complies with local regulations. Where LI is mandatory, they have to monitor different types of traffic (e.g., voice, data, or multimedia) and services (e.g., including multicast and broadcast, where the same traffic flow is directed at multiple subscribers). Furthermore, interception has to cover traffic generated by both the operators' on-network subscribers, and subscribers visiting (roaming) from other networks.

LI operations have to be implemented in a way that does not affect network performance for intercepted subscribers to ensure that subscribers are unaware they are being targeted. All network traffic has to be accessible to LI, even though only a very small fraction of these records are sent to the LEA. As a result, it is crucial for operators to minimize the impact on the overall network performance due to LI.

9.1 WiMAX® and Lawful Intercept

As a technology that combines a native IP core and support for mobility, WiMAX® presents significant lawful intercept challenges. In fixed or cellular voice networks, LI is relatively straightforward as the phone number uniquely identifies the subscriber. In IP-based WiMAX networks, traffic is user agnostic and this one-to-one relationship disappears.

WiMAX subscriber traffic is linked to IP addresses that typically are set for a given session, but change over time. Subscribers may be randomly assigned a pseudo-identity during the authentication process. To ensure effective traffic monitoring, the traffic linked to this pseudo-identity has to be identified.

As a result, LI requirements are more difficult to meet within WiMAX networks and, more generally, IP networks. Control data and session content may have to be intercepted at different locations within the network to ensure that all traffic associated with the targeted subscriber can be reliably attributed to that subscriber, regardless of application type, service type, or access location.

For additional information on broadband intercepts including which WiMAX network elements are responsible for each part of an intercept, Intercept Access Points (IAPs) and subject identification, see WiMAX Forum NALI Specification [68] section 4.2 and 5.

9.1.1 Requirements

An overview of the WiMAX aspects of LI is found in [67]. Specific requirements for packet data LI for North America is found in [68]. For IMS-based VoIP [70] and other basic Multimedia Services (those relying on a basic session start and end) in the U.S. [ATIS-0700005-2007 ""Alliance for Telecommunications Industry Solutions Technical Requirements for Lawfully Authorized Electronic Surveillance (LAES) for 3GPP IMS-based VoIP and other Multimedia Services.", 2007] should be referenced. Other regions identify the specifications to be used for IMS LI in their region. The WiMAX Forum® is developing IMS-specific specifications for Lawful Intercept of VoIP calls.

10 Emergency Calls

The deployment of commercial VoIP requires the support for emergency services (ES) (e.g. known as E9-1-1 in North America, 112 in the European Union). For ES, the Mobile Stations location information is needed in order to correctly route the emergency call to an appropriate emergency response center and to deploy an emergency response to the caller's location. Furthermore, to meet the stringent call set up requirements, the emergency service is better supported by the serving network (e.g. during roaming, it will be served by the visited network).

Rather than addressing the regulatory recommendations for Emergency services for all countries, some of the discussions below on the ES support may be more US centric.

The detailed WiMAX® network framework for providing support of Emergency Services can be found in WiMAX Forum Specification [70], Emergency Services, and the WiMAX Forum® IMS Specification [69]. This framework is built upon the overall WiMAX network architecture Release 1.0 and 1.5. It supports VoIP services provided over WiMAX networks by a VoIP service provider (VSP) that can either be part of an NSP or can be a third party commercial VSP with contractual relationship with the WiMAX NSP. The interfaces to existing ES provider network infrastructure (PSAPs) are identified, but the specification of such interfaces is beyond the scope of the white paper. The WiMAX ES framework supports communication between a VSP's VoIP server and the PSAP over both VoIP protocols or PSTN based protocols.

Although the exact regional or country-specific regulatory requirements that govern the support of ES for VoIP may vary, this white paper assumes that the basic ES capabilities would be universally applicable. It describes the framework for WiMAX networks to support ES for WiMAX commercial VoIP services. It provides an overall architectural framework for ES handling in WiMAX networks, and specifies all functionality that is required in the ASN/CSN for ES support.

10.1 FCC Enhanced 911 Requirements

Recently the U.S. Federal Communications Commission (FCC) has ruled that all (wired and wireless) VoIP services which interconnect with the public switched telephone networks (PSTN) must provide emergency operators with the locations and telephone numbers of callers [1]. Most of the other requirements of VoIP for the Voice Service Provider (VSP) over LAN also apply to VoIP over WiMAX networks. A VSP is defined as an entity that provides interconnected VoIP services to the WiMAX end user.

Per FCC requirements [1], VSPs are required to implement Enhanced 911 for the entire US within 120 days from the time VoIP services are deployed. VSPs are dependent on incumbent carriers or other third-party providers to support Enhanced 911. However, incumbent carriers have no obligation to cooperate with VSPs to provide Enhanced 911. VSPs must coordinate with incumbent carriers or other third-party providers to ensure Enhanced 911 requirements are met.

FCC requirements include:

1) Basic 911 requirement of call routing to the appropriate Public Safety Answering Point (PSAP);
2) Location Identification;
3) Determination of a call back number;
4) Handling of emergency calls during peak times;

It is recommended that any authenticated WiMAX device that supports VoIP have the ability to dial emergency numbers for non-subscribed WiMAX users. An additional recommendation is that WiMAX Devices have batteries to support 4 to 8 hours of operation in case of a power outage.
In this section, for each of the FCC requirements and above recommendations, general guidelines are provided based on the ecosystem as described in section 3 .

10.2 Location Identification Methods

For VoIP emergency call support, the location of the caller may need to be known for emergency services to be delivered. For PSTN this is known as the line is fixed to a location and with cellular, the location can be determined by GPS, assisted GPS or cell tower triangulation. IP telephony is packet-based and operates outside the traditional switching and transport elements of the PSTN. Furthermore, IP addresses and access modes are not fixed to a location. The portability complicates the emergency calls as IP based services in general are not mapped to a physical location [2]. Therefore, it is critical to develop location identification methods for IP telephony.

As part of NWG Rel 1.5, the WiMAX Forum has implemented and released the Location Based Services Specification [72], which supports both the commercial and emergency VoIP services. The following describes several methods for determining location for WiMAX.

10.2.1 Fixed Location

One method to determine a user's location is for the user to provide the operator with the street address where the WiMAX device is located. If the user changes the address where the device is located the operator would need to be informed. This is not an ideal solution as changes in address are not always communicated.

With this method, the end-user is responsible to notify the Operator when the address changes; however, the FCC requirements in the future may require the Operator to discover the end-user location. Operators may also use GPS to determine the location of the user. However, GPS may not be a common feature of fixed WiMAX devices. DHCP, ASN-GW, and serving BS ID and cell can assist in determining the area in which the user is located, but not the precise location. However, they can be helpful in determining whether the user has moved from the original address.

10.2.2 Network Triangulation

User location could be determined by triangulation. However, this method is not reliable, particularly in multi-path environments. Triangulation is particularly problematic in rural areas where a user is not likely to be within the coverage of multiple cells.

10.2.3 GPS

Although GPS is not available in all devices, it is ideal as it can provide the precise location of a user. GPS is commonly used for mobile and nomadic users, but can also be used for fixed services. Assisted-GPS further improves fast locking of GPS devices on cell phones and can also be utilized in WiMAX devices. In order not to deplete the battery, power-efficient methods for GPS can be employed [3]. The main challenge with GPS is that it requires the device to locate a signal with no less than three or four satellites (a.k.a. direct line of sight). This is typically not possible when the device is underground, deep in a building or tunnel, etc.

10.2.4 Combination of Methods

WiMAX devices can use any combination of the above described methods in the following order: Assisted GPS, GPS, Network Triangulation, core-network devices, and manual address entry mode for fixed devices. The performance can be further enhanced if all the methods are used to verify user's location.

10.3 Routing Emergency Calls

PSAPs receive emergency calls and route them to firefighting, police, and ambulance services. An emergency call made within a location normally requires the corresponding emergency service from that location. In order to properly route the emergency calls, the location identification methods described earlier can be used to locate the caller. It is important that the location of the caller is identified, otherwise the emergency services may not be delivered. For example in Canada, a Vonage customer residing in Toronto moved to Calgary and the emergency call they made in Calgary triggered an ambulance to go to an address in Toronto since the PSAP routing was not properly handled [6]. Several methods for routing calls are described above. Other methods may also be deployed.

10.4 Call-Back Number

Although leaving a call back number is not a requirement for Basic 911, FCC mandates a call back number for VoIP services with Enhanced 911. This requires WiMAX VoIP devices to have a call back number [1].

10.5 Priority During Peak Traffic Times

The delivery of voice traffic can be challenging during peak traffic times. Emergency calls during this time cannot be declined and need to be directed to PSAP's. WiMAX devices need to implement guaranteed QoS to ensure prioritization of emergency calls and to optimize the system between dedicated emergency calls and BW efficiencies [4].

10.6 Non-Subscribed WiMAX® Users

WiMAX service providers can enable calls to emergency services for non-subscribed users with a WiMAX VoIP client. This service is currently available to cell phone users where the phone is only given access to make emergency calls even if the phone has no credits to make regular calls.

10.7 Operation with Battery Support

Emergency cases may arise when there are power outages. If the user device needs a wall-outlet, then it will be useless unless it has battery power. Cable operators deploy Embedded Multimedia Terminal Adapters (EMTA) with battery support. For fixed WiMAX users, this is recommended. Nomadic and mobile users are assumed to have devices with batteries.

10.8 Local Breakout for Emergency Services (ES)

In roaming scenarios, it is highly desirable to provide local breakout for ES since only the local network may know the appropriate ES network that can handle the ES call in the user's roaming area.

Since ES is not subscription service, it can be supported entirely in the visited network without interaction with the home network. With local breakout for ES, the user continues to use the home network for all other subscription-based services.

Local breakout requires intelligence in the device to detect a native or roaming ES call and then select the visited network for the service. The MS can detect it is roaming through the NSP-ID advertisement information. In scenarios where more than on visited network is available, it is desirable for the MS to be able to selected a visited network based on the capabilities supported to avoid delays in attempting an ES call in one and then retrying in another if the first is not capable of providing local treatment for the call.

Once network selection is done, the MS initiates an ES based network entry (if necessary) to obtain a local IP-address and a local VoIP Server (e.g. WVS server or P-CSCF). If the visited network can support a local breakout, the visited AAA includes the visited network addresses (i.e. address of the local VoIP Server and DHCP server) in the request to the home AAA. The home AAA is configured with the ES profile, and based on the profile, the parameters in the request, it sends back in the response for the local VoIP server to be used for the ES call or the address of the local DHCP server provisioned with the local VoIP server's IP address. The MS then uses the DHCP Proxy/relay mechanisms to retrieve the local VoIP server's info.

If the MS does not have the capability of detecting an ES call or has roamed to a visited network which directs the call to the home network, the home SIP server/proxy should be able to indicate to the MS via a SIP 380 (alternative service) response that it should initiate the ES call in the visited network.

11 Priority Access for Emergency Telecommunications Service (ETS)

ETS is a national service, that enables authorized personnel to establish and maintain priority telecommunications (e.g., voice/video, data) using the public networks during network congestion conditions (ITU-T Study Group 2). Without ETS, the National Security and Emergency Preparedness personnel will suffer from session establishment blocking similar to that of the regular public users, and will not be able to carry out mission critical communications during natural and man-made disasters. The priority capabilities of ETS voice/video services are invoked at session set up on a session-by-session basis. The priority capabilities of ETS voice/video are revoked at the session disconnect. The ETS data transport priority service is invoked by the ETS user. After invocation, the data transport of all data services receives priority treatment until the user revokes the service.

The WiMAX Forum® is developing ETS specifications in two phases. Phase 1 ETS focused on priority indication and treatment for originating and terminating calls on WiMAX® Release 1.0, Release 1.5, and Release 1.6 networks based on IEEE Std 802.16-2009. Phase 2 ETS is being developed for WiMAX Release 2 networks with enhanced priority support for the IEEE Std 802.16m air interface and ETS voice/video services (e.g., use of SIP Resource Priority Header [IETF RFC 4412]).

12 Roaming

Roaming is a business and technical relationship, typically between two NSPs, that enables the subscribers of a Home NSP (H-NSP) to connect to and receive services through a second, or visited network. The services may include internet, e-mail, voice, video and other services available on the home network.

When a device is used by a WiMAX® Subscriber outside the home network coverage area, the device can connect to an access point of a different, or Visited NSP, as long as a valid roaming arrangement and appropriate service level agreement is set for VoIP are in place between the VNSP and the H-NSP.

The business components to make VoIP available in a roaming environment are as follows:

- The WiMAX Subscriber must have VoIP services provided by the H-NSP which extend to a roaming environment; and
- The V-NSP must have an agreement with the H-NSP to provide VoIP services or to enable routing back to the H-NSP for the WiMAX Subscriber to access VoIP services via the home network.

There are several complexities in enabling and providing VoIP roaming services. For example, if the H-NSP uses different classes of service with different user profiles to distinguish H-NSP WiMAX Subscribers which should be allowed access to VoIP services, then the H-NSP must be able to communicate this information to the V-NSP. Further, the V-NSP would need to have the capability of distinguishing between those H-NSP WiMAX Subscribers that should receive VoIP services and those that should not. There are also issues concerning providing QoS as having the capability of wholesale billing between two NSPs for premium services such as VoIP in a roaming environment. These issues are beyond the scope of this white paper. To avoid some of these issues initially, one option is to provide VoIP to all H-NSP customers based on any QoS parameters agreed between the V-NSP and the H-NSP. NSPs can implement billing arrangements directly or via third party WiMAX Roaming Exchanges.

When a WiMAX user is roaming, the VoIP platform may be located either in the visited network or the home network. Further, the platform may be on-premise equipment (of either the V-NSP or H-NSP) or a 3rd party hosted system through either the V-NSP or the H-NSP.

Conclusion

WiMAX® technology, as the first generation of 4G broadband wireless technology with built-in QoS support, provides cost-effective network solutions for WiMAX operators to offer ubiquitous VoIP services. By providing a WiMAX VoIP network, the WiMAX service provider can:

- Leverage the broadband capacity and built-in airlink QoS capability of WiMAX technology to deploy low-cost and high-quality converged voice and data services;
- Reduce both CapEx and OpEx while deploying a converged wireless and wireline IP network infrastructure for their 4G services;
- Extend their market share beyond the fixed or wireless subscribers to the Fixed Mobile Convergence (FMC);
- Compete with the incumbent the wireless cellular service providers;
- Outperform off-the-shelf best effort VoIP solutions; and
- Increase operator revenue by offering an additional service.

This white paper has highlighted the key network elements and components as well as WiMAX Forum standardized VoIP network solutions enabling the successful WiMAX VoIP ecosystem. It has also discussed the key technology challenges and design considerations and how WiMAX technology today can address these concerns.

This paper has presented key performance objectives for WiMAX operators to aim for high quality VoIP performance based on analysis that has been studied and recommended by the general WiMAX community paper.

Finally, the paper has presented essential supporting features, such as Location Based Service and inter-Radio Access Technology interworking and mobility support, enabling seamless VoIP services over WiMAX network, to provide WiMAX operators and vendors with a general understanding of the system requirements to support VoIP services over WiMAX networks.

Abbreviations

AAA	Authentication, Authorization and Accounting
AF	Application Function
AR	Access Router
ASN-GW	Access Service Network – Gateway
ASP	Application Service Provider
ATA	Analog Telephony Adapters
BE	Best Effort
BS	Base Station
BS ID	Base Station ID
BW	Bandwidth
CBR	Constant Bit Rate
CDR	Call Detail Record
CID	Channel Identifier
CODEC	Compression/Decompression
CPE	Customer Premise Equipment
CQICH	Chanel Quality Indication Channel
CSCF	Call Session Control Function (P(roxy), I(nterrogating), S(erving, E(mergency))
CSN	Connectivity Serving Network
CSR	Call Success Rate
DCR	Dropped Call Rate
DHCP	Dynamic Host Configuration Protocol
DiffServ	Differentiated Services
DL	Downlink
DSL	Digital Subscriber Line
DR	Dual Radio
E2E	End-to-end
EMTA	Embedded Multimedia Terminal Adapters
EPC	Evolved Packet Core
ePDG	Evolved Packet Data Gateway
ert-PS	Extended Real-Time Polling Service

EVDO	Evolved Data Optimized
FA	Foreign Agent
FCC	Federal Communications Commission
FDD	Frequency Division Duplex
FMC	Fixed Mobile Convergent
FTP	File Transfer Protocol
GERAN	GSM EDGE Radio Access Network
GPS	Global Positioning System
HARQ	Hybrid Automatically Repeat Request
HSR	Handoff Success Rate
HSS	Home Subscriber Server
HTTP	Hypertext Transfer Protocol
IAP	Intercept Access Point
IEEE	International Electrical and Electronic Engineer
IETF	Internet Engineering Task Force
IMS	Internet Multi-media Services
IntServ	Integrated Service
IP	Internet Protocol
ITU-T	International Telecommunication Union
IVR	Interactive Voice Response
KPI	Key Performance Indicators
LA	Location Agent
LBS	Location Based Services
LC	Location Client
LEA	Law Enforcement Agencies
LI	Lawful Intercept
LMA	Local Mobility Agent
LS	Location Server
MAP	Medium Access Protocol
MAC	Medium Access Control
MAG	Mobile Access Gateway
MIP	Mobile IP
MME	Mobility Management Entity
MRC	Monthly Recurring Charges
MRP	Mouth Reference Point

ms	Millisecond
MS	Mobile Subscriber
NAI	Network Access Identifier
NALI	North America Lawful Intercept
NAP	Network Access Provider
NAT	Network Address Translation
NRC	Non-Recurring Charges
nrt-PS	Non Real-Time Polling Service
NSP	Network Service Provider
OCS	Online Charging System
OFCS	Offline Charging System
OFDMA	Orthogonal Frequency-Division Multiple Access
OMA-DM	Open Mobile Alliance – Device Management
OSI	Open System Interconnection
OTA	Over The Air
PCC	Policy and Charging Control
PCEF	Policy and Charging Enforcement Function
PCRF	Policy and Charging Rule Function
PDF	Policy Distribution Function
PDFID	Packet Data Flow Identifier
PESQ	Perceptual Evaluation of Speech Quality
P-GW	Packet Gateway
PHS	Packet Header Suppression
PMIP	Proxy Mobile IP
PPS	Packets Per Second
PSAP	Public Safety Answering Point
PSTN	Public Switched Telephone Network
PTT	Push to talk
QoE	Quality of Experience
QoS	Quality of Service
RAN	Radio Access Network
RAT	Radio Access Technology
RF	Radio Frequency
RFC	Request For Comment
ROHC	Robust Header Compression

RTP	Real Time Protocol
rt-PS	Real-time Polling Services
SBC	Session Border Controller
SDFID	Service Data Flow Identifier
SFA	Service Flow Agent
SFF	Signaling Forwarding Function
S-GW	Serving Gateway
SGSN	Serving GPRS Support Node
SIP	Session Initiation Protocol
SPR	Subscription Profile Repository
SR	Single Radio
TDD	Time Division Duplex
TFTP	Trivial File Transfer Protocol
TOS	Types of Services
UDP	User Data Protocol
UE	User Equipment
UGS	Unsolicited Grant Service
UL	Uplink
UTRAN	UMTS Terrestrial Radio Access Network (E-URTAN – Enhanced URTRAN)
VBR	Variable Bit Rate
VLAN	Virtual Local Area Network
VoIP	Voice over Internet Protocol
VPN	Virtual Private Network
VSP	Voice Service Provider
WIF	Wi-Fi Interworking Function
WVS	WiMAX® Voice Services

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[74] DRAFT-T37-10-R016v01-A_WiMAX-Wi-Fi-IWK

[75] DRAFT-T37-11-R016v01-A_SR-IWK

[76] DRAFT-T37-004-R015v02-C_WiMAX-3GPP2-IWK

[77] DRAFT-T37-009-R015v01-A_ 3GPP-3GPP-EPS-IWK

[78] 3GPP TS 23.401, GPRS enhancements for E-UTRAN access (Release 9)

[79] 3GPP TS 23.402, Technical Specifications Group Services and System Aspects; Architecture enhancements for non-3GPP access (Release 9)

[80] 3GPP2 A.S0023-0 v1.0, Interoperability Specifications for HRPD Radio Access Network Interfaces and Interworking with WiMAX

[81] 3GPP2 X.S0058-0 v1.0_WiMAX-HRPD, Interworking: Core Network Aspects

Annex A

IMT-2000 – WiMAX Forum® Submission for Data and Voice capacity of OFDMA TDD WMAN [IMT-2000]

Table 0-1. System Parameters

Parameters	Indoor	OIP	Pedestrian	Vehicular
Number of Cells	19 (3 floors)	19		
Cell configuration	Omni			3-sector
Operating Frequency	2500 MHz			
Duplex	TDD			
Channel Bandwidth	5 MHz			
Cell/Sector Radius	20m	150m	250m	1.5km
BS-BS distance	35m	260m	430m	2.6km
Antenna Pattern	Omni			70° (-3 dB) with 20 dB front-to-back ratio
BS Height	N/A			15m above rooftop
BS Antenna Gain	2 dB	10 dB		13 dB
MS Antenna Gain	0 dB			
BS Maximum Transmit Power	10 dBm	20 dBm		30 dBm
MS Maximum Transmit Power	4 dBm	14 dBm		24 dBm
# of BS Tx/Rx Antenna	1Tx, 2Rx			
# of MS Tx/Rx Antenna	1Tx, 2Rx			
Receiver Type	MMSE			
MS Noise Figure	5 dB			
BS cable loss	2 dB			
Frequency Reuse	1			
Users/Sector	10			
Traffic Type	Data: Full Buffer (incl. 12-byte AES encryption overhead) Voice: G.729 8 kbps (45B VoIP packet including 3-byte header compression, 12-byte AES encryption, and 2-byte HARQ CRC overheads) with Voice Activity factor 0.5 (2-state Markov model)			
Channel Estimation	Dependent on the MS speed; degradation amounts to ~1 dB for low mobility (3 km/h) and ~2 dB for high mobility (120 km/h)			
Antenna Correlation	Spatial correlation factor equal to 0.5 at the MS			
PHY Abstraction	MIC-based			
Scheduler	Data: Proprietary Proportional Fair Voice: MLWDF			
Link Adaptation	Adaptive for initial transmissions, non-adaptive for retransmissions, 2-frame delay			

	feedback implemented
HARQ	Max 4 retransmissions, non-adaptive, synchronous
HARQ Operating PER	30%
Power Control	No power control on downlink, slow power control (10 Hz) on uplink
Coding	CTC
Data Symbols per Frame	Data: 24 for downlink, 12 for uplink (after excluding downlink and uplink overhead) Voice: 12 for downlink, 18 for uplink (after excluding downlink and uplink overhead)
DL/UL Partition	24:12 for data and 12:18 for voice

Table 0-2 summarizes the results for the Indoor, OIP, Pedestrian, and Vehicular deployment cases.

The user data throughput is defined with respect to the entire TDD frame, including both downlink and uplink sub-frames. Thus, it is calculated as the successfully delivered bits at the MAC Service Access Point (SAP) averaged over the entire simulation time, where the simulation time includes both downlink and uplink sub-frame times.

The downlink/uplink data spectral efficiency is defined with respect to the downlink/uplink sub-frame of the TDD frame respectively. Thus, it is calculated as the successfully delivered bits at the MAC SAP averaged over all users and the net downlink/uplink resources.

Voice capacity is given in terms of Erlangs/MHz/cell at 1% blocking. It is obtained from the maximum number of users that can be supported while system outage is <5% and the 5^{th}-percentile of average FER over all users is <3%. The system outage is defined as the fraction of users in outage. A user is in outage if the short-term FER (over each 2s interval) exceeds 1% for >15% of the 2s intervals.

Table 0-2. IP-OFDMA System Performance (5MHz TDD)

Deployment	Data DL (24 symbols)		Data UL (12 symbols)		Voice
	5th %ile user throughput (kbps)	Spectral Efficiency (bps/Hz/cell)	5th %ile user throughput (kbps)	Spectral Efficiency (bps/Hz/cell)	Spectral Efficiency in Erlang/MHz/cell (Erlang defined as a pair of DL/UL connections)
Indoor Office	180	1.37	79	0.84	15.8
OIP	113	1.27	74	0.80	13.2
Pedestrian Outdoor	137	1.27	83	0.82	13.8
Vehicular	115	2.62	71	2.00	25.2