From: Langtry, Colin
Sent: 03 November 2005 16:36
Subject: Request for information on access technologies to support IP applications over mobile systems

Please find attached a request for input to the next meeting of ITU-R Working Party 8F, seeking information on access technologies to support IP applications over mobile systems. I would appreciate it if you could forward this liaison to any interested groups in your organization.

Yours sincerely,

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NOTE: DOC version of attached document, including attachments, is available at:
http://ieee802.org/secmail/msg07443.html

INTERNATIONAL TELECOMMUNICATION UNION

RADIOCOMMUNICATION STUDY GROUPS

17TH MEETING OF WORKING PARTY 8F
HELSINKI, 12-19 OCTOBER 2005

Revision 1 to
Document 8F/TEMP/293-E
18 October 2005
English only

Working Party 8F

LIAISON STATEMENT TO EXTERNAL ORGANISATIONS ON THE DEVELOPMENT OF A PDNR ITU-R M.[IP CHAR] "KEY TECHNICAL AND OPERATIONAL REQUIREMENTS FOR ACCESS TECHNOLOGIES TO SUPPORT IP APPLICATIONS OVER MOBILE SYSTEMS" IN RESPONSE TO QUESTION ITU-R 223-1/8

Working Party (WP 8F) would like to thank External Organisations (EOs) for the information received at its Helsinki meeting that was helpful in progressing the work toward the development of a PDNR ITU-R M.[IP CHAR] ("Key technical and operational requirements for access technologies to support IP applications over mobile systems") in response to Question ITU-R 223-1/8. This PDNR will be developed in close co-operation with WP 8A.

At its meeting in Helsinki, WP 8F had developed a Workplan for M.[IP CHAR] that is provided for information in Attachment 1.

WP 8F also produced a preliminary draft composite document (see Attachment 2), trying to encompass, in a comprehensive manner, the material provided in the several input documents that were received. WP 8F has developed an overall structure of the document. Material of a generic nature would be candidate for inclusion in the main Annex, but material that describes specific implementation would be candidate for inclusion into an attachment to the Annex. It is not envisaged that this deliverable will include descriptions of specific systems. This composite document will be used as a basis for discussion at the next meeting of WP 8F (#18, 25 January – 1 February 2006).

WP 8F encourage EOs to provide input to its next meeting on the specific topics addressed in the attached composite document. EOs are kindly asked to submit information focused on these topics, providing material for the relevant sections of the attached document where the specific concepts are addressed. The information should be submitted by 16 january 2006 to the contact indicated below.

Contact: Colin LANGTRY
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Attachments: 2

ATTACHMENT 1

Source: Doc. 8F/TEMP/294

MICRO WORKPLAN FOR [IP CHAR]

Document Type :	TBD
SWG Chair:	Nicola Pio MAGNANI, Telecom Italia, Italy, E-mail: nicolapio.magnani@telecomitalia.it, Tel: +39 011 228 7089
Editor:	Hitoshi YOSHINO, NTT DoCoMo, Japan, E-mail: yosino@mlab.yrp.nttdocomo.co.jp, Tel: +81-468-40-3555
Focus for Scope and work:	The working document towards a PDNR [IP CHAR] delineates the essential technical and operational high-level characteristics required in mobile systems to support IP applications. It is not envisaged that this deliverable will include description of specific systems. Such a document could also be of help in ITU-R in identifying the need for the development of revisions of existing Recommendations/Reports, or of new deliverables.
Related documents:	See Section 3 of the PDNR
Milestones:	**Quebec (Meeting No. 16)** • Produce initial document structure • Send LS to WP 8A on decision on [IP CHAR] • Send LS to EOs for specific input on the initial document **Helsinki (Meeting No. 17)** • Produce micro work plan/milestones • Receive text from EOs and WP 8A • Review document structure • Start text preparation • Send LS to EOs on specific topics **Asia (Meeting No. 18)** • Finalize document structure and scope • Receive text from EOs • Continue detailed text preparation • Send LS to WP 8A on the updated document and milestones • Send LS to EOs on specific topics **France (Meeting No. 19)** • Review micro workplan/milestones • Receive text from EOs and WP 8A • Continue detailed text preparation • Send LS to EOs on specific topics **Denver (Meeting No. 20)** • Receive text from EOs • Continue detailed text preparation • Decide on the form of the deliverable • Send LS to WP 8A on the updated document and workplan/milestones • Send LS to EOs on specific topics **Geneva (Meeting No. 21)** • Receive text from EOs and WP 8A • Finalize the working document towards [IP CHAR] • Send the final document to WP 8A for subsequent submission to SG 8#8 If sufficient input material is received by WP8F#20, it may be possible to finalise [IP CHAR] by 2006, as indicated in Question ITU-R 223-1/8.

ATTACHMENT 2

Source: Doc. 8F/TEMP/301

JOINT WP8A/WP8F WORKING DOCUMENT TOWARD
A PDNR ITU-R M.[IP CHAR]

KEY TECHNICAL AND OPERATIONAL REQUIREMENTS FOR ACCESS TECHNOLOGIES TO SUPPORT IP APPLICATIONS OVER MOBILE SYSTEMS

(Question ITU-R 223-1/8)

1 Introduction

[Editor's note: WP 8A and WP8F have not determined yet whether this PDNR will be in the form of a Recommendation or a Report. It would be a Recommendation if it can adequately address requirements for mobile systems to support IP applications, while it would be a Report if it only describes the capabilities of the existing mobile systems to support IP applications.]

As IP applications are expected to grow and experience greater demand, the need to access the Internet via mobile communications systems will also increase. The essential technical and operational requirements for future IP applications should be discussed and explored.

IP applications over mobile systems may require additional technical considerations and special consideration should be placed on Multimedia services over mobile systems

2 Scope

This PDNR defines the essential technical and operational characteristics needed to support IP applications over mobile systems.

3 Related Recommendations and Reports

Recommendation ITU-R M.1079: Performance and quality of service requirements for IMT-2000 access networks

Draft new Recommendation ITU-R M.[IP_PERF_METHOD] - Methodology for deriving performance objectives and its optimization for IP packet [data) applications in the mobile-satellite service (Doc. 8/84).

Recommendation ITU-R S.1711 - Performance enhancements of transmission control protocol (TCP) over satellite networks (pre-published Doc. 4/43R1)

Recommendation ITU-R M.1645: Framework and overall objectives of the future development of IMT-2000 and systems beyond IMT-2000

[Other Recommendations may be listed as appropriate]

Draft new Report ITU-R F.[9B/FWA-IP+ATM] - Design techniques applicable to broadband fixed wireless access (FWA) systems conveying Internet protocol (IP) packets or asynchronous transfer mode (ATM) cells (Doc. 9/55).

4 Considerations

The ITU Radiocommunication Assembly,

considering

a) that Internet Protocol (IP) applications are experiencing and are expected to continue to experience high rates of growth globally;

b) that demands to access the Internet via mobile communications systems using IP are increasing rapidly;

c) that demands for global Multimedia services via mobile communications systems are also increasing rapidly;

d) that IP applications over mobile systems may require additional technical considerations and special consideration should be placed on Multimedia services over mobile systems;

e) the need to provide IP compatible services over mobile systems;

f) that IP applications over mobile systems are expected to support services compatible to wired-line internet services with cost-effectiveness;

g) that new innovative technologies supporting IP applications over mobile systems have been developed and are now being standardized,

recognising

a) Resolution 101 (Minneapolis, 1998) on Internet Protocol (IP)-based networks.

5 Notings

The ITU Radiocommunication Assembly,

noting

a) that standards organizations are working on such technologies.

6 Recommendations

The essential technical and operational high level characteristics needed to support IP applications over mobile systems are included in Annex A.

Annex: A

Abbreviations

AAA	Authentication, Authorization, and Accounting
BR	Border Router
CN	Correspondent Node. A peer with which a mobile station is communicating. A correspondent node may be either mobile or stationary.
DHCP	Dynamic Host Control Protocol
E2E QoS	End-to-End Quality of Service
HA	Home Agent
HAAA	Home AAA
HDB	Home Data Base
IETF	Internet Engineering Task Force
IMS	IP Multimedia Subsystem
LAC	Link Access Control
MAC	Media Access Control
MMD	Multi-Media Domain
MS	Mobile Station
PDF	Policy Decision Function
PDSN/AGW	Packet Data Serving Node/ Access Gateway
PDU	Protocol Data Unit
PPP	Point-to-Point Protocol
SLA	Service Level Agreement
VAAA	Visited AAA
VDB	Visited Data Base
P-CSCF	Proxy-Call Session Control Function
PDSN	Packet Data Serving Node
S-CSCF	Serving-Call Session Control Function
RAN	Radio Access Network

Annex A

1 Introduction

With the high demand for Internet Protocol (IP) applications growing consistently at such high rates, the need to develop more efficient systems able to support newer internet features will be a necessity.

It is anticipated that users' demands for mobile services are gradually getting higher and more diverse and complex with increase of mobile communication market. Diverse and complex services are expected to have quite different traffic characteristics and QoS levels comparing to voice traffic or text. In addition, increase of demands on multimedia services causes further more complexity in designing and developing the future mobile systems. In other words, future mobile technologies should support diverse services with different traffic characteristics which are pervasive regardless of air interface technologies. In this sense, IP applications can be widely accepted as one of solutions affordable to provide compatible services since IP has been already used in wired communication systems. Therefore, IP applications over mobile systems will be important in the future due to global interoperability.

IP applications over mobile systems could be supported by existing standards which use IP to send data. However, to support more enhanced IP applications over mobile systems such as seamless delivery of multimedia data services, several significant technical characteristics in radio interface and access networks should be considered.

The AIPN shall be a common IP-based network providing common capabilities independent to the type of service being provided and the access system being used. Convergence to IP technology within the AIPN system design shall be considered from the perspective of the system as a whole with minimum duplication of functionality.

The AIPN shall be capable of accommodating a variety of different access systems hence providing a multi-access system environment to the user. The AIPN shall support service provision and provide mobility functionality within and across the different access systems.

An AIPN shall be able to accommodate fixed access systems and to inter-work with fixed networks in order to provide seamless services over fixed/mobile converged networks, e.g. charging and providing supplementary services.

An AIPN shall be able to inter-work with a variety of wireless broadband networks based on IP technologies.

The AIPN shall be under the control of the operator of the AIPN. The AIPN shall provide common mechanisms for the AIPN operator to control access to and usage of AIPN resources.

The AIPN shall provide a high level of basic system performance including low communication delay, low connection set-up time and high communication quality.

The AIPN shall provide efficient usage of system resources. In particular, the scarcity of the radio resource shall be respected within the AIPN by ensuring that radio resources are utilised as efficiently as possible. The AIPN shall also support effective usage of power resources within mobile terminals by minimising the impact on mobile terminal battery life (standby and active).

The AIPN shall be able to accommodate a vast number of terminals and users as well as be able to support a wide variety of diverse devices. Examples of terminals that shall be supported by the AIPN include terminals which main purpose is to include a sensor/RF tag, household appliances/media players with a wireless communication module, as well as traditional mobile terminals. Identification, addressing, and routing schemes within the AIPN shall be provided to support this communication environment; in particular the AIPN shall support naming and addressing schemes for a given user/session.

The AIPN shall be able to efficiently support a variety of traffic models e.g. user-to-user, user-to-multicast and traffic models generated by ubiquitous services.

The AIPN shall provide functionality as appropriate to support international roaming with other AIPNs.

The AIPN shall provide appropriate mechanisms to support independent operation of services, AIPN, and/or access systems.

During the initial stages of AIPN introduction it is likely that earlier systems and terminals will exist in parallel with AIPNs. Therefore, the AIPN should be able to support CS terminals and accommodate access systems based on CS. Interworking and interconnection with CS networks shall also be provided.

The AIPN shall be designed to enable efficient coexistence with earlier PS domains.

2 Basic capabilities

2.1 Introduction

Some essential technical characteristics necessary to support IP applications over mobile systems include:

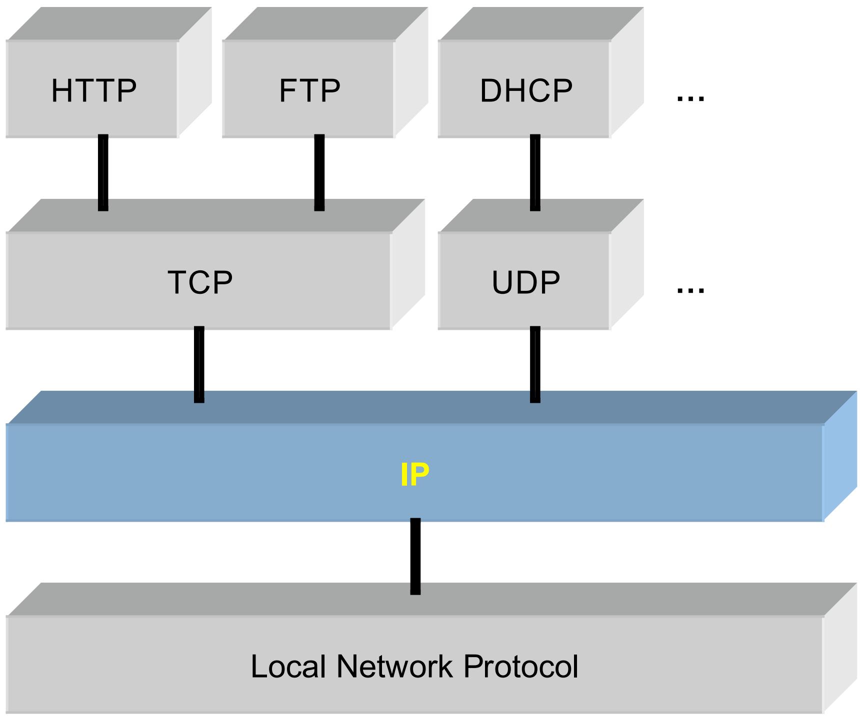
- Compatibility
- Transparency
- Scalability and efficiency
- Security
- High speed burst traffic
- Low cost per bits
- Seamless Delivery Services
- Various Grade of Services.

Mobile IP applications must be able to remain compatible to all levels used for the standard non-mobile IP. Also, IP over mobile applications should remain “invisible” for higher level protocols and applications. Higher layers should continue to function normally even while the mobile has altered its point of attachment to the network. As equally as newer features are developed, the subnetwork should also function with maximum efficiency at a minimum complexity.

As the rates of mobile communications increase, IP over mobile applications should be scalable over the large numbers utilizing the Internet. These subnetworks should also be able to provide support for such advanced internet features as Multicasting and Quality of Service (QoS). Identification, authentication and authorization are required in order to protect against remote attacks. As information is transmitted from node to node, the location of a mobile node must be authenticated to maintain security.

2.2 Encapsulation

The Internet Protocol version 4, IPv4, is described in IETF RFC 791. Internet Protocol version 6, IPv6, is described in IETF RFC 2460. The following Figure illustrates the protocol hierarchy.



All IP application related data is exchanged between hosts in IP packets. For example, the Hypertext Transfer Protocol [IETF RFC 2068] protocol data unit, PDU, is encapsulated in a Transmission Control Protocol [IETF RFC 793] PDU, which is encapsulated in an IP PDU, and then finally in a local network PDU. In the context of mobile systems, the local network may be considered the radio interface.

2.3 Address Management

To participate effectively in an IP network, a device must possess an IP address. IP packets are routed based on the destination address field within the packet header.

IPv4 addresses are 32-bit binary numbers. Most commonly, they are expressed by considering the 32 bits as four octets, converting each octet to decimal, then separating these numbers with dots to create dotted decimal notation. For example, the address 17.112.152.32 is the dotted decimal notation of:

00010001 01110000 10011000 00100000

IPv6 addresses are 128-bit binary numbers. If these addresses are expresses in human readable form they are most commonly expressed in hexadecimal with colons to aid legibility, for example:

fe80:0000:0000:0000:020a:95ff:fef3:2e91

2.4 Routing

[Editor's Note: Information to be Added]

2.5 Maximum Transmission Units (MTUs) and IP Fragmentation

IPv4 packets (datagrams) vary in size, from 20 bytes (the size of the IPv4 header alone) to a maximum of 65535 bytes. Subnetworks need not support maximum-sized (64KB) IP packets, as IP provides a scheme that breaks packets that are too large for a given subnetwork into fragments that travel as independent IP packets and are reassembled at the destination. The maximum packet size supported by a subnetwork is known as its Maximum Transmission Unit (MTU).

2.6 Multicasting

The Internet model includes "multicasting", where IP packets are sent to all the members of a multicast group. Multicast is an option in IPv4, but a standard feature of IPv6. IPv4 multicast is currently used by multimedia, teleconferencing, gaming, and file distribution (web, peer-to-peer sharing) applications, as well as by some key network and host protocols (e.g., RIPv2, OSPF, NTP). IPv6 additionally relies on multicast for network configuration (DHCP-like autoconfiguration) and link-layer address discovery (replacing ARP). In the case of IPv6, this can allow autoconfiguration and address discovery to span across routers, whereas the IPv4 broadcast-based services cannot without ad-hoc router support.

2.7 Bandwidth on Demand (BoD) Subnets

Some subnets allow a number of subnet nodes to share a channel efficiently by assigning transmission opportunities dynamically. Transmission opportunities are requested by a subnet node when it has packets to send. The subnet schedules and grants transmission opportunities sufficient to allow the transmitting subnet node to send one or more packets (or packet fragments). These subnets are referred to as Bandwidth on Demand (BoD) subnets. Examples of BoD subnets include Demand Assignment Multiple Access (DAMA) satellite and terrestrial wireless networks, IEEE 802.11 point coordination function (PCF) mode, and DOCSIS.

2.8 Bandwidth Asymmetries

Some subnetworks may provide asymmetric bandwidth (or may cause TCP packet flows to experience asymmetry in the capacity) and the Internet protocol suite will generally still work fine. However, there is a case when such a scenario reduces TCP performance. Since TCP data segments are "clocked" out by returning acknowledgments, TCP senders are limited by the rate at which ACKs can be returned [BPK98]. Therefore, when the ratio of the available capacity of the Internet path carrying the data to the bandwidth of the return path of the acknowledgments is too large, the slow return of the ACKs directly impacts performance. Since ACKs are generally smaller than data segments, TCP can tolerate some asymmetry, but as a general rule, designers of subnetworks should be aware that subnetworks with significant asymmetry can result in reduced performance, unless issues are taken to mitigate this [RFC3449].

Several strategies have been identified for reducing the impact of asymmetry of the network path between two TCP end hosts, e.g., [RFC3449]. These techniques attempt to reduce the number of ACKs transmitted over the return path (low bandwidth channel) by changes at the end host(s), and/or by modification of subnetwork packet forwarding. While these solutions may mitigate the performance issues caused by asymmetric subnetworks, they do have associated cost and may have other implications. A fuller discussion of strategies and their implications is provided in [RFC3449].

3 Quality-of-Service (QoS) Considerations

Some of the general requirements for QoS should include:

- End-to-end QoS
- Wide range of QoS-enabled services
- End-to-end QoS should be supported within the local domain and across different network domains as well
- Appropriate levels of QoS should be maintained even when internet features, such as multicasting is applied.

The QoS network architecture should be flexible enough to support different QoS control mechanisms, which are defined in different access technology environments. The network should be able to provide mechanisms which could perform traffic and congestion control. Traffic control would include functions such as network resource management, packet marking, traffic shaping, and packet scheduling. Congestion control refers to functions which include packet discarding or explicit congestion notification. The network should also provide methods in which to negotiate QoS at both transport and service layers, and allow dynamic alterations of the QoS parameters. The network should allow operators to implement QoS policy control, where the policy-based management would be extended across multiple domains to ensure QoS.[NGN]

It is generally recognized that specific service guarantees are needed to support real-time multimedia, toll-quality telephony, and other performance-critical applications[1]. For some important services with strict end-to-end QoS requirements, such as conversational speech or streaming video, the QoS shall be assured in case of integrated networking with different IP network domains or backbone networks. Most likely this would be ensured on a per service basis of specific flows of IP packets having been identified by the service.

There are at least two architectural approaches to providing mechanisms for QoS support in the Internet. The IP Integrated Services (Intserv) [RFC1633] is a working group formed to standardize a new resource allocation architecture and new service models for the internet. The Intserv model includes two main components: the traffic control and the ReSerVation Protocol. Flows are identified by a flow specification (flowspec), which creates a stateful association between individual packets by matching fields in the packet header. Capacity is reserved for the flow, and appropriate traffic conditioning and scheduling is installed in routers along the path. The ReSerVation Protocol (RSVP) [RFC2205] [RFC2210] is usually, but need not necessarily be, used to install the flow QoS state. RSVP is a control signaling protocol which requires the introduction of states for specific information flows, although reservation states are “soft” in that they are regularly renewed by messages sent from the initiator of the reservation request. If not renewed, the reservations are timed-out. [Source: 3GPP]
Intserv defines two services, in addition to the Default (best effort) service.

1 Guaranteed Service (GS) [RFC2212] offers hard upper bounds on delay to flows that conform to a traffic specification (TSpec). It uses a fluid-flow model to relate the TSpec and reserved bandwidth (RSpec) to variable delay. Non-conforming packets are forwarded on a best-effort basis.

2 Controlled Load Service (CLS) [RFC2211] offers delay and packet loss equivalent to that of an unloaded network to flows that conform to a TSpec, but no hard bounds. Non-conforming packets are forwarded on a best-effort basis.

The other architectural approach is called the IP Differentiated Services (Diffserv) [RFC2475]. This is an alternative resource allocation scheme, which provides service differentiation by dividing the traffic into different classes at the edge of a network by nodes classified as boundary nodes. A boundary node classifies each incoming packet into a particular traffic class. Diffserv provides a scalable approach, but it in itself does not provide guaranteed QoS. Diffserv does not make any per-flow reservations. Rather, it provides QoS for aggregates of flow. Resources are assured by the prioritization for traffic classes.

Mobile systems face inherent tradeoffs between delay, throughput, reliability, and cost. Some subnetworks have parameters that manage bandwidth, internal connection state, and the like.

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[1] Recommendation ITU-R M.1079 “Performance and quality of service requirements for IMT-2000 access networks”

Therefore, the following subnetwork capabilities may be desirable, although some might be trivial or moot if the subnet is a dedicated point-to-point link.

1 The subnetwork should have the ability to reserve bandwidth for a connection or flow and schedule packets accordingly.

2 Bandwidth reservations should be based on a one- or two-token bucket model, depending on whether the service is intended to support constant-rate or bursty traffic.

3 If a connection or flow does not use its reserved bandwidth at a given time, the unused bandwidth should be available for other flows.

4 Packets in excess of a connection or flow's agreed rate should be forwarded as best-effort or discarded, depending on the service offered by the subnet to the IP layer.

5 If a subnet contains error control mechanisms (retransmission and/or FEC), it should be possible for the IP layer to influence the inherent tradeoffs between uncorrected errors, packet losses, and delay. These capabilities at the subnet/IP layer service boundary correspond to selection of more or less error control and/or to selection of particular error control mechanisms within the subnetwork.

6 The subnet layer should know, and be able to inform the IP layer, how much fixed delay and delay jitter it offers for a flow or connection. If the Intserv model is used, the delay jitter component may be best expressed in terms of the TSpec/RSpec model described in [RFC2212].

7 Support of the Diffserv class selectors [RFC2474] suggests that the subnet might consider mechanisms that support priorities.

4 End to end requirements for AIPN with access system

4.1 Introduction

The AIPN with access system shall be capable of fulfilling certain end-to-end characteristics requirements. These requirements, if met by the system, will enable good end-user performance for a variety of end-user services including, but not limited to:

– Real-time, interactive applications, e.g., voice, video and real-time gaming applications

– Non-real time, interactive applications, e.g., Web browsing, remote login, chat

– Media streaming applications

– Conversational services.

The present requirement applies to the entire end-to-end path from the user terminal to the other end-host or server including radio network, core network, and backbone network processing, buffering and propagation delays. Since the delay depends on the location of the end-host, the requirements shall be considered as preferred values when the user is within the same continent.

4.2 Delay Characteristics

The TCP sender bases its retransmission timeout (RTO) on measurements of the round trip delay experienced by previous packets. This allows TCP to adapt automatically to the very wide range of delays found on the Internet. If the path delay variance is high, TCP sets an RTO that is much larger than the mean of the measured delays. If the packet loss rate is low, the large RTO is of little consequence, as timeouts occur only rarely. Conversely, if the path delay variance is low, then TCP recovers quickly from lost packets; again, the algorithm works well. However, when delay variance and the packet loss rate are both high, these algorithms perform poorly, especially when the mean delay is also high.

Because TCP uses returning acknowledgments as a "clock" to time the transmission of additional data, excessively high delays (even if the delay variance is low) also affect TCP's ability to fully utilize a high-speed transmission pipe. It also slows the recovery of lost packets, even when delay variance is small.

Mobile systems should therefore minimize all three parameters (delay, delay variance, and packet loss) as much as possible. Often these parameters are inherently in conflict. For example, on a mobile radio channel, retransmission (ARQ) and/or forward error correction (FEC) can be used to trade off delay, delay variance, and packet loss in an effort to improve TCP performance. While ARQ increases delay variance, FEC does not. However, FEC (especially when combined with interleaving) often increases mean delay, even on good channels where ARQ retransmissions are not needed and ARQ would not increase either the delay or the delay variance.

The tradeoffs among these error control mechanisms and their interactions with TCP can be quite complex, and are the subject of much ongoing research. It is therefore recommend that mobile systems provide as much flexibility as possible in the implementation of these mechanisms, and provide access to them as discussed above in the section on Quality of Service.

4.3 Round-trip delay requirement:

– The round-trip delay requirement for small IP packets (0 payload) shall be defined such that it will enable real-time applications at good quality when both end-hosts reside within the same continent.

– Acceptable value is 50-70 ms average round-trip delay.

4.4 Packet loss requirement:

– The packet loss at the IP layer requirement shall be defined such that it will enable a certain maximum download data rate for TCP based applications when both end-hosts reside within the same continent and the radio conditions are preferable.

– Acceptable value is 0.001% in good radio conditions but maximum 0.1% packet loss ratio.

NOTE – The maximum value for packet loss may also be sufficient for UDP applications.

4.5 Delay jitter requirement:

– The delay jitter requirement shall be defined such that it will not harm real-time applications and will not cause significant TCP degradation due to spurious timeouts.

– Acceptable value for delay jitter is 25 ms (for real-time gaming), for TCP based download applications occasional bursts of up to 50 ms are still tolerated.

5 Supporting Mobility

5.1 Introduction

Internet users are increasingly mobile. Not only are many Internet nodes laptop computers, but pocket organizers and mobile embedded systems are also becoming nodes on the Internet. These nodes may connect to many different access points on the Internet over time, and they expect this to be largely transparent to their activities. Except when they are not connected to the Internet at all, and for performance differences when they are connected, they expect that everything will "just work" regardless of their current Internet attachment point or local subnetwork technology. Changing a host's Internet attachment point involves one or more of the following steps.

First, if use of the local subnetwork is restricted, the user's credentials must be verified and access granted. There are many ways to do this. A trivial example would be an "Internet cafe" that grants physical access to the subnetwork for a fee. Subnetworks may implement technical access controls of their own. It is common practice for both cellular telephone and Internet service providers (ISPs) to agree to serve one anothers' users; RADIUS [RFC2865] is the standard method for ISPs to exchange authorization information.

Second, the host may have to be reconfigured with IP parameters appropriate for the local subnetwork. This usually includes setting an IP address, default router, and domain name system (DNS) servers.

On multiple-access networks, the Dynamic Host Configuration Protocol (DHCP) [RFC2131] is almost universally used for this purpose. On PPP links, these functions are performed by the IP Control Protocol (IPCP) [RFC1332].

Third, traffic destined for the mobile host must be routed to its current location. This roaming function is the most common meaning of the term "Internet mobility".

Internet mobility can be provided at any of several layers in the Internet protocol stack, and there is ongoing debate as to which is the most appropriate and efficient. Mobility is already a feature of certain application layer protocols; the Post Office Protocol (POP) [RFC1939] and the Internet Message Access Protocol (IMAP) [RFC3501] were created specifically to provide mobility in the receipt of electronic mail.

Mobility can also be provided at the IP layer [RFC3344]. This mechanism provides greater transparency, viz., IP addresses that remain fixed as the nodes move, but at the cost of potentially significant network overhead and increased delay because of the sub- optimal network routing and tunneling involved.

Some subnetworks may provide internal mobility, transparent to IP, as a feature of their own internal routing mechanisms. To the extent that these simplify routing at the IP layer, reduce the need for mechanisms like Mobile IP, or exploit mechanisms unique to the subnetwork, this is generally desirable. This is especially true when the subnetwork covers a relatively small geographic area and the users move rapidly between the attachment points within that area. Examples of internal mobility schemes include Ethernet switching and intra-system handoff in cellular telephony.

However, if the subnetwork is physically large and connects to other parts of the Internet at multiple geographic points, care should be taken to optimize the wide-area routing of packets between nodes on the external Internet and nodes on the subnet. This is generally done with "nearest exit" routing strategies. Because a given subnetwork may be unaware of the actual physical location of a destination on another subnetwork, it simply routes packets bound for the other subnetwork to the nearest router between the two. This implies some awareness of IP addressing and routing within the subnetwork. The subnetwork may wish to use IP routing internally for wide area routing and restrict subnetwork-specific routing to constrained geographic areas where the effects of suboptimal routing are minimized.

The AIPN shall support end-user, terminal and session mobility.

An AIPN shall be capable of providing seamless terminal mobility within and across access systems. The user shall experience no disruption in the service due to terminal mobility.

An AIPN shall be capable of maintaining a service during a change in access system, with no perceivable interruption from a user perspective.

An AIPN shall support adaptation of services to the capabilities provided by the access systems during terminal mobility.

An AIPN shall support terminal mobility based on criteria including radio conditions, service requirements, user preferences and operator policies.

In cases when there is a degradation in service quality due to terminal mobility it shall be possible to notify end users of the degradation.

The AIPN shall provide appropriate mechanisms to enable users to connect to the AIPN through multiple access systems.

The AIPN shall be capable of supporting handover between CS voice services and AIPN equivalent services (e.g. Voice over IP).

5.2 Mobility & Handover

The system shall support mobility across the cellular network and should be optimized for low mobile speed from 0 to 15 km/h. Higher mobile speed between 15 and 120 km/h should be supported with high performance. Mobility across the cellular network shall be maintained at speeds from 120 km/h to 350 km/h (or even up to 500 km/h depending on the frequency band). Voice and other real-time services supported in the CS domain in previous systems shall be supported via the PS domain with at least equal quality as supported by previous systems (e.g. in terms of guaranteed bit rate) over the whole of the speed range.

The impact of intra system handovers on quality (e.g. interruption time) shall be less than or equal to that provided by CS domain handovers in previous systems.

The mobile speed above 250 km/h represents special case, such as high speed train environment. In such case a special scenario applies for issues such as mobility solutions and channel models. For the physical layer parameterization the system should be able to maintain the connection up to 350 km/h, or even up to 500 km/h depending on the frequency band.

The system shall also support techniques and mechanisms to optimize delay and packet loss during intra system handover.

6 Security Requirements

6.1 Introduction

There are several security issues related to supporting IP applications whether the applications are run over a mobile system or not. Charging for an application often requires the user to be authenticated to ensure the user is allowed to access the application. In addition, the user may also need to authenticate the application to ensure that personal details, e.g. user's current location or bank details, are not revealed to the wrong application. Many applications will also have the need to encrypt and integrity protected the data that is exchanged between the mobile and application. This ensures that the data sent between the mobile and application can not be read by a man-in-the-middle or modified without it being detected by the receiving party.

The following general objectives should be considered in IP applications over mobile systems:

– Confidentiality
– Integrity
– Accountability
– Availability
– Non-repudiation
– Privacy

6.2 Authentication

[Editor's Note: Information to be Added.]

6.3 Privacy

[Editor's Note: Information to be Added.]

6.4 Lawful Intercept

[Editor's Note: Information to be Added.]

7 Service Architecture

The service architecture provides a secure, extensible framework under network operator control. It supports efficient, flexible deployment of end-to-end IP applications. This architecture includes infrastructure elements for service processing, subscriber databases, media gateways and servers, and nodes for policy and charging control, as well as the terminals themselves. Real-time multimedia IP services may be provided as well as less challenging services like streaming and interactive services like messaging.

8 Interoperability and Interworking

8.1 Introduction

[Editor's Note: Information to be Added.]

8.2 Legacy Systems

[Editor's Note: Information to be Added.]

9 Transmission Efficiency

The cost and limited availability of spectrum requires that the wireless technology transporting the IP traffic be as efficient as possible.

At the physical layer the measure of spectral efficiency is bits per second/Hz of spectrum used. Optimizing this requires a careful design of the system to minimize overhead and exploit the different QoS requirements of IP applications, the variability of the wireless channel, and the mobility of multiple users in the network.

Above the physical layer, the spectral efficiency for IP applications is maintained by using protocols with low overhead such as compression protocols. In particular, IP header compression protocols for IP multimedia such as RObust Header Compression, specified in IETF RFC 3095.

10 Example of Multi-media Applications

The wide range of multimedia applications that run over IP networks requires that the wireless system be designed to support all the particular requirements of each application.

10.1 Voice-over-IP (VoIP) over Mobile

Following are the advantages of carrying voice over IP versus conventional circuit-switched voice services:

1 The packetization of voice allows the voice codec and network to jointly exploit variations in the amount of information in the speech signal to reduce the data rate that is needed while not sacrificing voice quality. This can translate into increased network voice capacity.

2 The flexibility of the IP transport and intelligent-client architecture of IP networks allows the timely introduction of rich and new supplementary services that can be used with the voice services.

In the future, as the use of VoIP applications becomes more and more widespread, the enablement of voice services based on wireless IP will be essential.

VoIP services over mobile systems will have to overcome the following challenges:

- Good QoS support over several different network technologies
- Real-time service insists strict delay and packet loss requirements over packet switched networks
- Factors Influencing Speech Quality:
  - Impairment of audio quality due to audio processing
  - Delay, Jitter, and Loss of packets
  - Wireless environment guarantees fluctuations in transmission quality
  - Physical layer leads to highly error-prone environment
  - Wireless links generally offer lower bandwidth compared to wireline links
  - Packet losses due to bit errors
- Mean Opinion Score (MOS)

10.2 Web surfing

The web-surfing experience is no longer limited to static webpages described by markup languages. With the increasing use of downloaded executable applets and scripts, the web surfing multimedia experience has expanded beyond still images to dynamic animation. The richness of these downloaded applications requires downloading of larger clips for execution in the user's web browser.

10.3 Push-to-talk

[Editor's Note: Information to be Added.]

10.4 Interactive Audio and Video

[Editor's Note: Information to be Added.]

10.5 Streaming Audio and Video

Streaming audio applications generally require medium amounts of bandwidth with a bound on the amount of total jitter experienced at the receiver. Streaming video requires higher bandwidth with similar bounds on the jitter.

Since streaming applications are generally not real time, the transport of streaming traffic typically does not require immediate delivery. However, to limit the size of the de-jitter buffer required in mobile terminals and reduce tune-in delay for broadcast streaming services, the jitter must be bound by the network.

10.6 Digital Video Telephony

Video Telephony is one of the most challenging IP applications to support since it requires high bandwidth, low latency, and low jitter.

11 Conclusions

[Editor's Note: Information to be Added.]

Appendices

Specific applications are illustrated in six Attachments.

Attachments: 6

Attachment 1 to Annex A

Source: Document 8A/214, 8A/254

Description of some specific ip applications over mobile systems

1 Introduction

Section 2 contains information on features currently available in 3GPP, whereas Section 3 deals with future developments currently under study.

2 Features currently available

2.1 IP Multimedia Subsystem (IMS)

In Release 5, IMS architecture has been developed in order to provide multimedia services based on IETF developed protocols SIP/SDP and provide service enablers in order to provide services developed within and outside of 3GPP (e.g. enable 3rd party service integration with IMS). The architecture for IMS enables delivery of services like speech, video, combinational services, presence, IMS messaging etc. IMS also provides the infrastructure for easy service development; for example, Push to Talk over Cellular (PoC) is one example of the usage of IMS in order to deliver a specific service and ability to charge for these services independent of PS domain charging.

Additional functions that were developed during Release 6 over IMS are necessary enablers to support PoC, IMS messaging, and Presence services. By combining the support of messaging with other IMS service capabilities, such as Presence, new rich and enhanced messaging services for the end users can be created such as Instant Messaging, Chat, Store and Forward Messaging with rich multimedia components. The concept of presence, whereby users make themselves "visible" or not "visible" to other parties of their choice, allowing services such as group and private "chats" to take place. Presence is an attribute related to mobility information, and provides a different capability to be exploited by other services. The concept of presence enables other multimedia services to exploit this key enabler to support other advanced multimedia services and communications. See also TS 23.228 – “IP Multimedia Subsystem (IMS); Stage 2”.

2.2 End to end QoS procedures and Service based Local Policy control

In Release 5, end to end QoS architecture for UMTS networks and interworking with external network procedures have been developed over the PS domain, where interworking may be achieved by:

– signalling along the flow path (e.g. RSVP, LDP).

– packet marking or labelling along the flow path (e.g. DiffServ, MPLS)

– interaction between Policy Control and/or Resource Management elements.

– Service Level Agreements enforced by the border routers between networks.

Service based local policy provides mechanism for binding media, policy control over the media (and UMTS bearer) based on the session information from IMS signalling, charging correlation and gating function .

2.3 Support for IPv6

Cooperation with IETF has resulted achieving improved support of IPv6 PDP contexts in the PS domain.

2.4 IP Transport in UTRAN

In Release 99 and Release 4, only ATM can be used at the transport layer in the various interfaces. This Work Item introduces the possibility to use IP at the transport layer in the Iub, Iur, Iu-Ps and Iu-Cs interfaces, as an alternative to ATM. However, the use of ATM at the link layer under IP is not precluded.

The introduction of IP as a transport protocol in the radio network does not imply an end to end IP network; the UE may be given an IP address by the higher layers, but it will not be part of the UTRAN IP network (which is private), and packets will be encapsulated in the corresponding User Plane protocol.

The Work Item has made a choice for the protocols to transport the Radio and Signalling bearers over IP. Different solutions are adopted: UDP is used in the User plane in the three interfaces, and SCTP with additional protocols is used for the Signalling bearers. With respect to the IP version, IPv6 is mandatory and IPv4 is optional, although a dual stack is recommended.

Additionally, the Work Item resulted in decisions on QoS and interworking with ATM transport networks:

– Diffserv is the mechanism to provide different service levels, and several alternatives are allowed for the traffic flow classification. It is allowed also that the QoS differentiation can be provided either on a hop-by-hop basis or on a edge-to-edge basis;

– Interworking with Release 99/Release 4 and Release 5 ATM nodes is required, and it can be accomplished via a dual stack, a logical interworking function or a separate InterWorking unit.

2.5 RObust Header Compression functionality (ROHC)

Under the Release 4, the RObust Header Compression functionality (ROHC) was introduced. Its benefit is an important reduction in header overhead, simply because the fields of the headers of IP packets are either constant or changing in a known pattern. Hence it is possible to send only information regarding the nature of the changing fields of those headers. This leads to a reduction in the total size of header+payload, from 60 octets into 20 octets for some applications (e.g. IP based voice applications) and with IP version4, and from about 80 octets to 20 octets with IP version6. This translates directly into bandwidth efficiency. The ROHC scheme is claimed to be more suited to cellular environment and changing links than the previous compression schemes. RFC3095 "RObust Header Compression (ROHC)" is the IETF proposal for IP header compression specially designed for real time IP services over wireless links. ROHC was included in the Release 4 of UTRAN as one of the compression schemes to be provided by the PDCP (Packet Data Convergence Protocol) sublayer in the RNC. As ROHC is part of the PDCP layer, there is a compressor and decompressor pair in the RNC and a corresponding pair in the UE. During SRNS relocation the source RNC gives the role of the serving RNC (SRNC) to the target RNC, therefore compressor/decompressor have to be relocated as well. The straightforward solution in place in Release 4 was to initialise the header compression in both peers after relocation, which results in problems like high probability of lost speech frames. This could be avoided by not initialising compression but continuing it in the target SRNC from the place in which the compression ended in the source SRNC. The required changes to perform RFC3095 context relocation in the SRNS context relocation were introduced in Release 5. In order to perform the ROHC relocation, RANAP messages that carry RAB contexts during SRNS relocation are updated to carry also the ROHC/RFC3095 contexts for each RAB. The ROHC context IE to be transferred is defined in the RRC protocol specification. "RFC3095 Context Info" container to RANAP information elements "Forward SRNS Context" and "RANAP Relocation Information" were added to RANAP. Further details can be found in TR 25.844 ("Radio Access Bearer Support Enhancements") and in TR 25.860 ("Radio Access Bearer Support Enhancements") in TR 25.844 - "Radio Access Bearer Support Enhancements" and TR 25.860 - "Radio Access Bearer Support Enhancements" respectively.

2.6 IMS Signalling Flag

IMS signalling traffic is carried by an interactive PS RAB. In order to enable UTRAN to apply special handling of "signalling RABs" compared to other interactive PS RABs, a signalling "flag" was introduced as an additional level of QoS for Interactive traffic RABs.

2.7 High Speed Downlink Packet Access (HSDPA) and Enhanced Uplink

As enhancements of the support for packet data transmission in Release 99/Release 4, the Release 5 radio-interface specification includes enhanced features for High-Speed Downlink Packet Access (HSDPA), allowing for highly efficient downlink packet-data transmission with peak data rates up to 14 Mbit/s and simultaneous high-speed packet data and other services such as speech on the single carrier. Furthermore, the Release 6 radio-interface specification includes features for Enhanced Uplink access allowing for improved capacity and coverage, data rates up to more than 4 Mbps, and uplink radio-interface delay less than 10 ms.

2.8 Multimedia Broadcast and Multicast Services (MBMS)

The Release 6 radio-interface specification includes a radio access network architecture providing efficient support for Multimedia Broadcast and Multicast Services, i.e. allowing for multimedia content distribution to groups of users over a point-to-multipoint bearer.

2.9 Flow Based Charging

A new architecture has been developed in order to support Flow Based Charging, allowing more granular and flexible charging principles and mechanism for PS domain, this is applicable to both On line and Off line charging cases and allows operators to activate/deactivate charging rules according to their own policies. This function uses IETF developed Diameter protocols and provides support of such functions as (see also TS 23.125 – "Overall high level functionality and architecture impacts of flow based charging; Stage 2"):

– Identification of the service data flows that need to be charged individually (e.g. at different rates), and those that can be handled as an aggregate;
– Provision and control of charging rules on service data flow level;
– In the GPRS case: Provision and control of charging rules on a per PDP context basis;
– Reporting of service data flow level byte counts (for volume based charging) and service data flow durations (for time based charging);
– Event indication according to on-line charging procedures (e.g. sending AAA Accounting Stop) and, optionally, following this particular event, taking appropriate actions on service data flow(s) according to the termination action.
– Event indication and event monitoring and following this particular event, taking the appropriate on-line charging actions.

2.10 WLAN-UMTS interworking

Interworking of WLAN enables 3GPP–WLAN Interworking so as to extend 3GPP services and functionality to the WLAN access environment. The 3GPP–WLAN Interworking System provides bearer services allowing a 3GPP subscriber to use a WLAN to access 3GPP PS based services. The following functionalities have been specified:

- Provide the interworking WLAN with a means of Access, Authentication and Authorisation (AAA) through the 3GPP System, which allows WLAN UEs to access WLAN and the locally connected IP network (e.g. Internet)
- Provide WLAN UEs with IP bearer capability to access PS based services which are provided by PLMN.

For further details, see also TS 23.234 – "3GPP system to Wireles Local Area Network (WLAN) interworking; System description".

3 Future developments

With enhancements already incorporated in the 3GPP Specifications, the 3GPP radio-access technology will be highly competitive for several years. However, to ensure competitiveness in an even longer time frame, i.e. for the next 10 years and beyond, a long-term evolution of the 3GPP radio-access technology needs to be considered. This is the scope of the Study Item on Evolved UTRA and UTRAN: the SI sheet can be found in SI sheet for "Evolved UTRA and UTRAN". Currently two TR are being developed: one on the feasibility study for Evolved UTRA and UTRAN and the second one focusing on the requirements for Evolved UTRA and UTRAN. Within this framework a number of advanced techniques will be considered; the study of some of those has been already initiated (e.g., OFDM - see SI sheet for "Analysis of OFDM for UTRAN enhancement").

IMS is considered to be crucial for the development of multimedia-based 3G networks. In order to make the deployment of IP based multimedia services economically viable in a 3G environment, it is necessary to ensure that the Radio Access Bearers used to support these services are optimised. RABs for IMS support are already defined in 3GPP UTRAN Rel5. However, these RABs may need to be optimised, in order to ensure a commercially viable deployment of IMS services. Work is ongoing in the scope of the Radio Access Bearer Support Enhancements WI, to analyse UTRAN Rel5 and look at different optimisation proposals to improve the support of IMS in Release 6 or later. The work have been focused on VoIP specifically, since it is where the optimisation is most needed when comparing a non optimised IMS speech call and a R99 CS speech call.

Currently work has either started or will be starting on the following aspects of architecture development:

All-IP Network (AIPN): The AIPN is a common IP-based network that provides IP-based network control and IP transport. This includes the provision of IP-based mobility control of the high quality appropriate for cellular networks (i.e. no degradation in performance compared to other cellular mobility mechanisms) that is not dependent upon specific access or transport technologies, or IP version. It is the aim of the AIPN to provide a seamless user experience for all services within and across the various access systems. As well as across multiple diverse terminals a user may possess. Interworking with external IP networks (e.g. Internet) and legacy networks (e.g. PSTN) is provided and functionality at the edge of the network enables support of different access systems and legacy equipment. See also SI sheet for "All-IP Network" and TR 22.978 – "All-IP network (AIPN) feasibility study".

SMS/MMS over generic 3GPP IP access: The overall objective is to enhance the 3GPP specifications to support delivery of SMS and MMS over WLAN and any other 3GPP IP access in a manner which guarantees existing SMS and MMS services are not degraded. See also TR 23.804 – "Support of SMS and MMS over generic 3GPP IP access".

Combinational services: Many operators regard IMS as a key feature. However, there remain issues with the efficiency of transferring Voice over IP over the radio interface, and, with the capability of the GSM radio interface to handle VoIP. Additionally, operators are interested in techniques to smooth the rollout and accelerate the take-up of IMS. As a result of this, a study was started to study the techniques for delivering IMS services using CS bearers for real-time media components. The study shall cover the different solutions for offering existing IMS simultaneous services (real time media + non real -time media) especially in GERAN, where conversation PS spectrum efficiency is too low. See also TR 23.899 – "Combining Circuit Switched (CS) bearers with IP Multimedia Subsystem (IMS)".

Evolution of Policy Control and Charging: Goal is to provide a full harmonization of various policy and charging functions that have been developed within 3GPP. Such harmonization is essential when optimizing real-time interactions of the GGSN (and gateways of other IP Connectivity Access Networks), and optimizing the real-time control architecture of GPRS in general. Goal is also to study how differentiation based on end-user subscription classes can be achieved. In addition it should be studied how non-QoS policy control functions (e.g. service authorization, control of redirect functions etc.) fits in the harmonized architecture. These aspects are important in order to fully capitalize on the new core network capabilities that have been developed. See also TR 23.803 – "Evolution of policy control and charging".

System Enhancements for Fixed Broadband Access to IMS: The standardization of the Next Generation Network (NGN) is addressed by a number of SDOs, e.g. ETSI and ITU-T. 3GPP recognises that external standards organisations are in the process of defining NGN session control using IMS as a platform. This will embed IMS as the framework for advanced services for many types of operators. It is expected that some enhancements of the 3GPP specifications will be needed for IMS to meet the NGN requirements.

3GPP System Architecture Evolution: The objective of this feasibility study is to develop a framework for an evolution or migration of the 3GPP system for a higher-data-rate, lower-latency, packet-optimized, multi-RAT, access technology. The focus of this work will be on the PS domain with the assumption that voice services are supported in this domain. The main objectives is to address the following aspects:

– Overall architecture impacts stemming from activities requirements developed from TSG-RAN Study Item on Radio Evolution. The architectural developments should take into account the targets for the evolution of the radio-interface,

– Overall architecture impacts stemming from the work in SA1 on an All-IP Network (AIPN) (see SI sheet for "All-IP Network" and TR 22.978),

– Overall architecture aspects of supporting mobility between heterogeneous access networks, including service continuity.

Migration aspects should be taken into account for the above, i.e. how to migrate from the existing to or evolve to any new architecture. See also TR 22.978 – "All-IP network (AIPN) feasibility study".

Architectural Enhancements for End-to-End Quality of Service (QoS): The work investigates possible solutions to enhance the end-to-end QoS architecture as currently specified in 3GPP TS 23.207 to achieve improved end-to-end QoS in the case of interworking with IP network domains or backbone networks that provide IP QoS mechanisms and enhanced interworking with other next generation networks. Within this study, emerging QoS standardization efforts from TISPAN, ITU-T, and the IETF should be taken into account. See also TR 23.802 – “Architectural enhancements for end-to-end Quality of Service (QoS)”.

3GPP will contribute further on these topics, also reflecting the progress on the activities currently ongoing.

Attachment 2 to Annex A

Source: Document 8A/237

LOCATION INFORMATION SERVICES FOR MOBILE INTERNET SYSTEMS

1 Abstract

As the mature stage of wireless voice communication services, new data services based on Internet should be developed to expand the market and provide technical benefit to users. Especially, mobile communication services are very essential to moving users, and location specific data services could be the killer applications of wireless Internet services.

In this contribution, new emerging technology of location information service is explained, and this could be the good guideline or recommendation for service providers and manufacturers to provide valuable mobile communication services to users.

2 Introduction

IP (Internet Protocol) is used for delivering datagram from source to destination, and there are three methods for the delivering. The first one is unicasting, which delivers datagram to the destination according to the IP address in the header of datagram. Currently, it is known that the unicasting is the unique method for delivering user data to destination, even for broadcasting services. Thus, as the total number of users to access the server increases, the rapidly growing load to the server and the network traffic of Internet degrades network performance and QoS (Quality of Service). The next one is multicasting, which is the proper method for Internet broadcasting service. But it needs seamless multicasting routers in Internet, and it is not available in public. Thus for Internet broadcasting services, the same datagram should be copied so many times, and delivered from a server to every terminal, and this causes traffic congestion problem to the network and the server. To resolve this problem, RMCP (Relayed MultiCasting Protocol), which combines unicasting and multicasting was proposed, and standardized in ITU-T/JTC1. It employed two kinds of Internet delivery method of unicasting and multicasting. At first, the broadcasting server sends the broadcasting datagram to a remote router by unicasting method, and next the router distributes the broadcasting datagram to terminals by multicasting method. This tries to overcome the lack of multicasting routers, but it still needs multicasting routers with this new function at the edge network, and somewhat complex multicasting protocol. The last Internet delivery method is broadcasting, which causes too much traffic congestion to the network, so it couldn't be used for user data delivery in Internet. It has been just used for special purposes inside subnet.

For the point of user's convenience, Internet protocol has some limitations. All transmitting and receiving nodes must have their unique IP addresses for unicasting delivery of datagram. But it is impossible to assign unique IP addresses to all the Internet devices such as personal computers, PDAs (Personal Digital Assistance), and mobile phones in the world, with current IPv4 protocol. Thus the protocols of temporal assignment of IP address, such as DHCP(Dynamic Host Configuration Protocol), should be employed, and specifically for moving nodes the management of the IP address can not be avoidable to coincide the IP address of the node to the IP address of corresponding subnet. The usability and the convenience of Internet services are very limited especially in mobile computing environment. On the other hand, all the Internet services are provided by client-server basis. From terminal, user should access the service server and retrieve the data that user want. Thus anyone, who wants to receive Internet services, must know the corresponding IP address or the domain name of the service server, prior to Internet access. In this case, if the user doesn't know the domain name or it was changed, there should be additional manipulation to find out it by web search engine. This is very difficult job for pedestrians or drivers with portable terminals.

As the market of voice communications saturated, network providers are actively seeking new business area, and one of the most prominent services is LBS(Location Based Services). While pedestrians or drivers are passing through an area, location specific information is provided to them without pre-knowledge of web site address or web searching and managing IP address. But due to the reasons mentioned above, existing Internet protocol is not proper solution to provide Internet local information services. For these kinds of LBS applications, IP local information technology is the best solution, to provide the information with convenience and usefulness, even without user's position information. Recently, IMT-2000 system of 3GPP uses cell broadcasting technology to broadcast data into all mobile terminals in a cell. It can provide very efficient local information technology with peculiar protocol, not Internet protocol. In 4G mobile system which pursues all IP network, IP protocol should be supported from server to terminals.

In this contribution, a new Internet service scheme is proposed for local information services by Internet protocol, and this technology can be applied to all the communications and the broadcasting networks which use Internet as core network.

3 Location Information Technology for Mobile Internet

According to Internet protocol, for the datagram with destination IP address of all 1s, they are supposed to be delivered to all of the terminals in the subnet. Although this method has not been used for data broadcasting services, Internet data broadcasting inside the subnet is possible. In this case, IP datagram broadcast to the subnet has nothing to do with local subnet IP address or IP addresses of terminals. All of the terminals in the subnet can receive the IP datagram, without any pre-settings, such as IP address, subnet mask, gateway address, domain name server address, web site address, and so on. When the IP datagram is broadcast, the destination address of data link layer also has to be set up as broadcasting address. For wireless LAN, the 48 bits of MAC (Media Access Control) address should be all 1s. Then IP data broadcasting services in a subnet can be provided by existing Internet protocol, but not through the Internet. To expand the IP data broadcasting service through whole Internet, there should be additional method to carry the broadcasting datagram into the subnet. If the unicasting delivery of Internet datagram from server to subnet, and the broadcasting in subnet are combined, then the proposed technology has great synergy, especially for IP local information services. Thus, amendment of the function to the Internet equipment in an edge network is necessary, which is destination IP address translation.

FIGURE 1

The network configuration of IP local information system

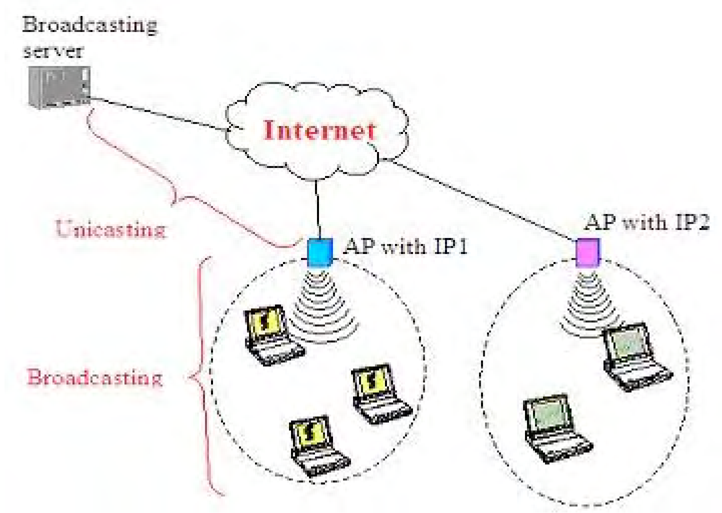
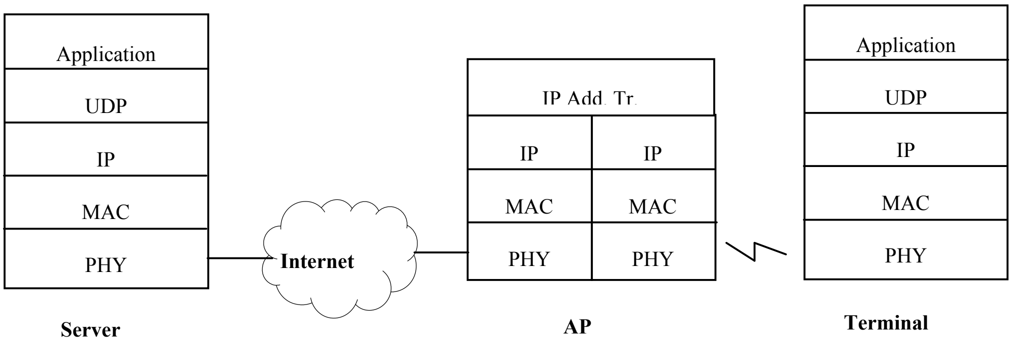


Fig.1 is the network configuration of IP local information service system for wireless LAN. Wireless LAN network likes hot-spot has been actively deployed worldwide, and almost all of the AP(Access Point)s are connected to Internet by Ethernet or xDSL (x-Digital Subscriber Line). APs of ISP(Internet Service Provider) have their own unique IP addresses for operation and management purposes. For IP local information services, this IP address means the position of the AP, and broadcasting server has the list of IP addresses and corresponding position information of APs. At first, the broadcasting server sends the Internet datagram of broadcasting to an AP with unique destination IP address corresponding to specific position by unicasting method. If the AP detects the datagram of matching IP address, next scans protocol number and port number for distinguishing broadcasting datagram from the other datagram. For broadcasting services, UDP (User Datagram Protocol) should be used (i.e., The protocol number is 17.) instead of TCP (Transport Control Protocol), and port number should be defined for this application, for example, 3333, among server, APs, and terminals. For the broadcasting datagram, the AP changes the destination IP address of the datagram to all 1s, and also sets up MAC address to all 1s. Finally, the AP transmits the MAC frame to air by radio signal. At the terminal side, all the associated wireless LAN receivers in the radio zone of the AP receive the frame, and it is bypassed to IP layer. In addition, the IP layer bypasses the datagram to UDP layer without any IP address management of the terminal, due to the IP address of all 1s. Then finally, the broadcast data frame is to be delivered to the viewer program of the terminal, according to the pre-defined port number. The user can receive broadcasting data or find the information related to the user position, without any positioning equipment or pre-knowledge of web site address. Fig. 2 is the protocol stack of IP local information system.

FIGURE 2

The protocol stack of IP local information system



Location information services could be provided to users without position information of users, with local portal technology. When a user is getting into a communication zone, through authentication process by the ISP, it can recognize the position of the user. So the ISP can redirect the web page of the user's terminal into the local portal corresponding to the user position. This uses not broadcasting technology but unicasting. The comparison among existing multicasting, local portal, and proposed IP local information system is shown in Table 1. There are advantages and disadvantages for the methods according to Internet services, but the proposed technology is the best solution for Internet local information services. It provides local information to the users with very efficient and convenient way by expense of simple functional addition to AP.

TABLE 1

Comparison between existing and IP local information methods

Item	Multicasting	Local portal	IP Local Information
Network equipment function	Multicasting routers	Nothing special	IP add. Translation
Multicasting registration of terminal	Need	No	No
Pre-knowledge of web site address	Need	No	No
Unique IP address for terminal	Need	Need	No
Location based Information	No	Yes	Yes
Position information of user	Need	No	No
Number of users	Limited	Limited	No limit
Repetitive broadcasting	No	No	Yes
Network & server traffic	Least	Much	A little

Repetitive Broadcasting and Data Filtering

As a user moves in the radio zone, the data receiving time is limited according to the size of the zone and the velocity of the moving user. For a user to listen valuable data once at least in the zone, it should be broadcast repetitively in the zone. This could be easily implemented by employing data storage unit in the AP. In order to broadcast repetitively, there should be the control field in the header of broadcasting data, and it includes priority, period, and duration information for the repetition. When the AP receives the broadcasting data, it first analyzes the control field, and stores them into the data storage unit. According to the value of the field, the AP retrieves the data from the storage unit for repetitive transmitting. On the other hand, the IP local information data sent from service server to APs, would be very huge and their contents would be versatile. If they all were displayed at the terminal, it would be very difficult for user to identify the data of concern from them. By allocating the type of data field in the header of broadcasting data, user can display only the necessary data to the terminal. Simple packet structure of broadcasting data is shown in Fig.3. As user turns on a common radio receiver and tunes dial for listening to specific station, in IP local information system the terminal is displayed only the specific data by user’s setting of the viewer program of the terminal. As the broadcasting data is not requested by the terminal but pushed into the terminal, user doesn’t have to know the web site address of broadcasting station, and just select the preferred data from receiving data. If the displayed data is written by markup language, like HTML (Hyper Text Markup Language), the user can easily move to corresponding web site by just clicking the received text. This is the true convergence of broadcasting and communications, that is, one-way broadcasting services and interactive communication services through unique communication network.

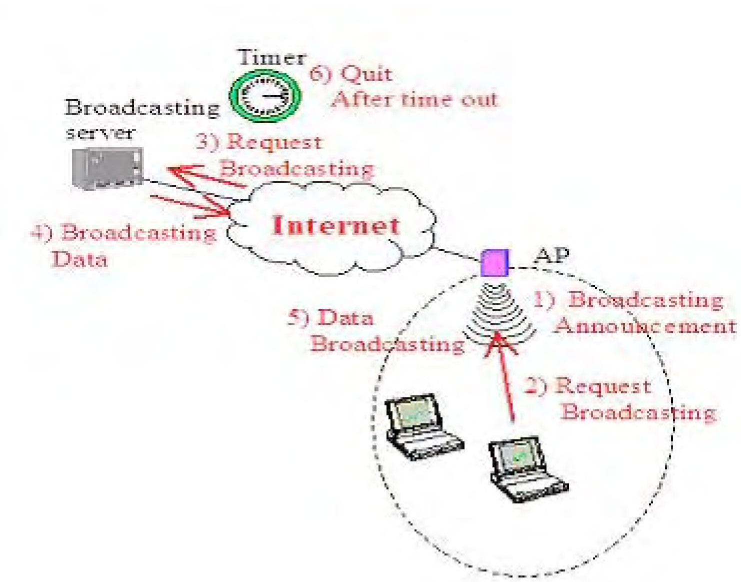
FIGURE 3

The header format of IP local information data frame

Control field	Type of data field	Broadcast data

FIGURE 4

The modified functions of IP local information system for terminal triggering

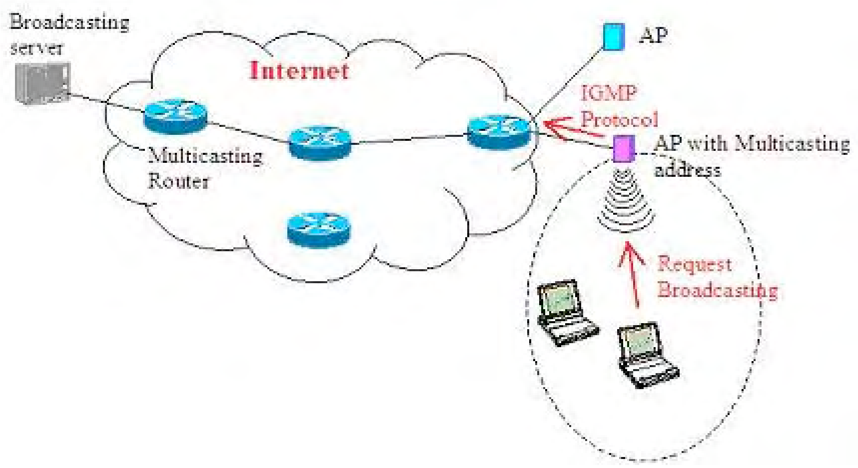


Triggering for Local Information Service

The proposed IP local information technology is very simple and easy to use, but there would be some inefficiency. Even there is no user in radio zone, the server always sends the data through the Internet. In order to get rid of the inefficiency, terminal triggering for the IP local information method is proposed. Also. Fig. 4 shows the modified functions of IP local information system for terminal triggering. AP broadcasts announcement message repetitively into the radio zone. When a terminal gets into the radio zone, it could receive the message from the AP. Then the terminal transmits broadcasting request datagram to the AP, with destination IP address of all 1s, UDP protocol, and pre-defined port number, for example, 3333. So the AP can receive and recognize the triggering datagram from the terminal. Next, the AP sends another broadcasting request datagram to server. When the server receives triggering datagram, it starts on timer and sends broadcasting data to the AP, until the timer is out. If there is no trigger datagram from the AP until timeout, the server ceases to send the broadcasting data. Thus, the terminal that wants to receive broadcasting data has to send the trigger datagram to the AP repetitively. This method seems like multicasting protocol, but this is much simpler than multicasting, and doesn't need multicasting address and mobility management. By employing triggering datagram with some complexity, the unnecessary broadcasting traffic is removed.

FIGURE 5

The most efficient IP local information system using Multicasting



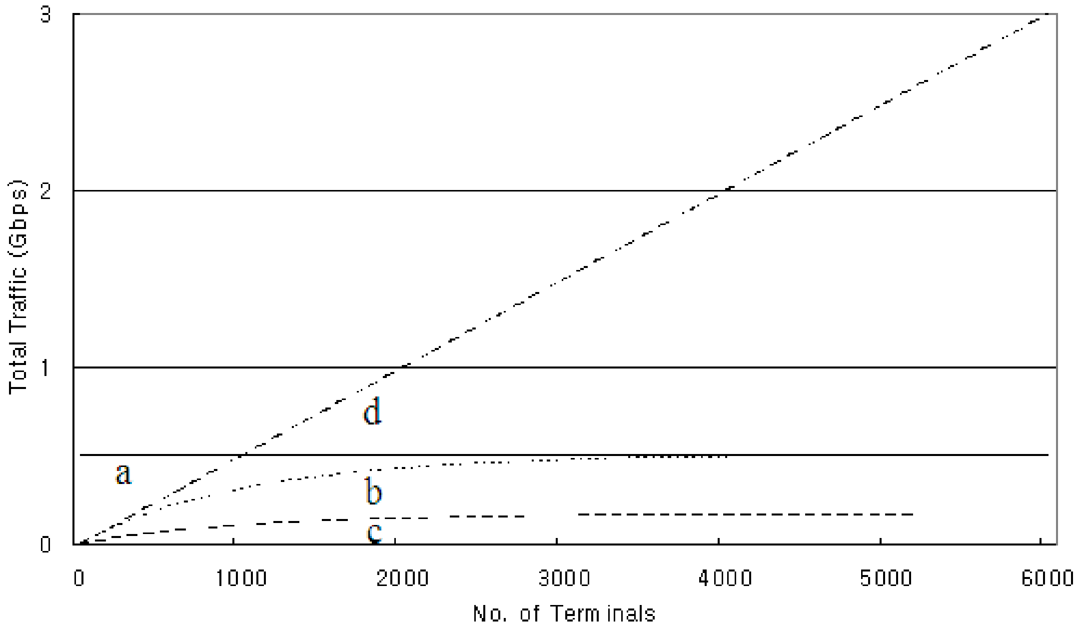
Multicasting + Broadcasting

IP local information method is the only available method for local information, if the multicasting is not possible due to the lack of multicasting routers. But as time goes by, the legacy network equipment would be replaced by new equipment that supports IPv6 and multicasting function. Even for the time, still IP local information system has peculiar advantages, as shown in Table 1. For several APs connected to a same router, multicasting is more efficient than unicasting in the point of traffic. Multicasting between broadcasting server and an AP and broadcasting between the AP and terminals is the best way to provide IP local information services, as in Fig. 5. When the AP receives the triggering datagram from the terminal, it registers multicasting process to Internet by IGMP(Internet Group Management Protocol). Thus, the information data from the server is to be delivered just once to the edge router, not twice as in unicasting method. In this case, the same information data is broadcast to the two APs. This is the most efficient for data traffic with easy manipulation rather than existing multicasting method.

Fig. 6 shows network traffic comparison for existing method and proposed methods. This is the simulation result for the case of a broadcasting server, thousand of APs connected to Internet, and a number of terminals associated to the APs are in uniform distribution to the each AP. In addition, it is assumed that three APs in average are connected to an edge router. The broadcasting traffic from the server to each AP is assumed as 500kbps. As a result, for existing unicasting method, the total network traffic is increased linearly as the number of terminals increasing. In contrast that, the total traffic is fixed according to the number of APs, not depend on the number of terminals, for IP local information system. On the other hand, the trigger method gets rid of the waste of traffic, and multicasting merged with broadcasting method reduces the network traffic inversely proportional to the number of APs connected to a same router. As the protocol becomes complex, the traffic efficiency is increased.

FIGURE 6

Network and server traffic comparison for existing method and proposed methods. (a: pure IP local information system, b: IP local information with trigger, c: IP local information with multicasting and trigger, d: existing unicasting method)



4 Location Based Services using Location Information Technology

IP local information technology is very useful to provide local information based on the position of each AP. Moreover this technology can be used for general broadcasting services, such as Internet television, Internet radio, IP datacasting, and so on. Thus, every user in a radio zone can receive broadcasting contents with or without payment through Internet.

FIGURE 7

The perfect convergence between communication and broadcasting

FIGURE 8

The LBS applications using local information technology

LBS applications

- Local Geographic Information Service
- Local Community News Service
- Local Shopping Information Service
- Local Tourist Information Service
- Emergency Guide Service
- Traffic Information Service

As user gets into the radio zone, the user can receive location specific information without any position information of user. By employing battery and data storage unit in AP, evacuation information could be announced for a while without network connection and power supply. In addition, due to the storage unit, repetitive data broadcasting is possible without server and network traffic load.

Private Broadcasting Services

- Announcement Service
- e-Education Service
- Conference Service

FIGURE 9

The conference service

When so many people gather into a place and need common information, local information broadcasting is the best solution. However, due to the lack of frequency band, it is very difficult to get the license of broadcasting station from radio administration. Instead of licensed station, wireless LAN network with IP local information system could be unlicensed broadcasting system. If an AP located at the center of stadium and it uses omni-directional antenna, then it can cover full range of stadium and support all the visitors, because the technology doesn't need unique IP address.

5 Conclusion

The proposed technology in this contribution overcomes the limit of existing Internet services, and provide the convergence between broadcasting and communications. Furthermore, user convenience is highly upgraded, and service providers and network administrators can easily configure the local information network, due to the reduction of traffic of Internet and server, and burden of IP address management. This could be very valuable solution to LBS (Location Based Service), Telematics and ITS(Intelligent Transport Service) applications that are mainly for mobile users. This proposed technology was tested well using wireless LAN AP and notebook computer, of which Windows is the operating system. It shows the possibility that broadcasting datagram could be used for Internet services, and it could create new service paradigm of Internet application.

Attachment 3 to Annex A

Source: Document 8A/267

A Protocol for improving UDP performance over wireless networks and its application to mobile robot control

1 Abstract

New emerging wireless internet (WiBro in Korea) era can make many new services. The intelligent mobile robot service is the one of new services. The Wibro can control the robot remotely. In the control, both the on-line realtime and the reliability of control data transmission are very important. Considering the realtime control and data reliability, new protocol for the robot service is designed. It is described that a UDP protocol with some policies is effective for the mobile robot service. Two flow charts for packets transmission are showed. Finally it is commented for both the interval time decision making of local retransmitting and another protocol policy for moving users.

2 A control Protocol for mobile robot over wireless internet

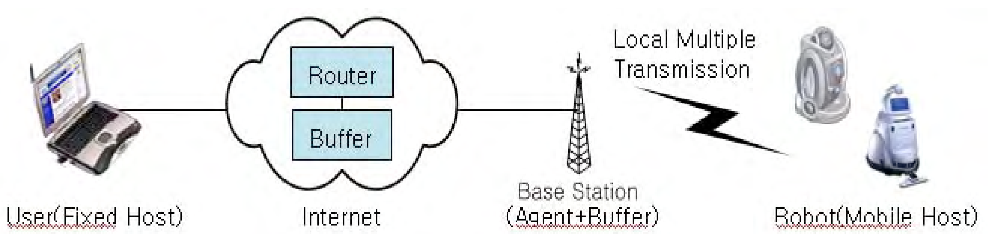
When one controls the remote mobile robot through wireless internet, we must consider on line realtime and reliability of robot control data transmission. There are two major protocols in internet, TCP (Transmission Control Protocol) and UDP (User Datagram Protocol). In TCP, the protocol divides these data into multiple packets and transmits. After confirming the arrival of the data (receiving the ACK packet), TCP transmits the next packet. The TCP protocol uses the time more than UDP. Because the TCP is particularly targeted at the wired networks, a packet loss is assumed to be caused by the network congestion. In the wireless environment, the chances to lose packets transmission bit errors are not negligible. Especially there are prone to bursty packet error. Therefore the TCP protocol is not proper for application to realtime control of robot.

We select the protocol UDP for robot realtime control through wireless internet. The UDP also have some problem. UDP is prone to error even though the protocol is speedy. In this contribution, to overcome the problem, we adopt some policies (algorithms) under UDP. Both robot control data transmission under prorioty condition in wired links and using multiple transmissions in wireless links can make the mobile robot act on time rightly.

Fig.1 shows the network environment consisting of wired links and wireless links. The user can control remotely the mobile over wireless internet.

FIGURE 1

The protocol of network and mobile robot



The robot control data and links are defined as follows:

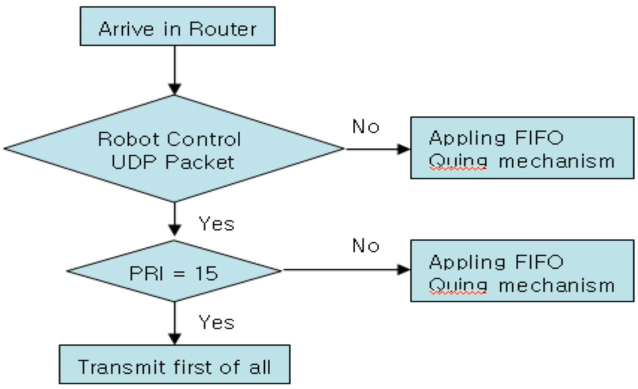
Robot control data : Simple command to order the robot move or do some works. Example) key inputs on mobile phone panel, key inputs on computer keyboard

Wired links : To accomplish the real time transmission of the UDP packet for robot control and overcome the congestion packet loss, set the priority field(PRI) of IP packet as 15(Ipv6). The node (router) serves the arrived packet of robot control data with PRI 15 first of all.

Wireless links : To exclude the transmission congestion loss of robot control data in wireless links(which have essential difficulties), copy the robot control UDP packet at Base Station and transmit multiply to robot. The length of wireless link is usually shorter than wired links. So the delay time of wireless links is of brief duration than transmission time of wired links.

FIGURE 2

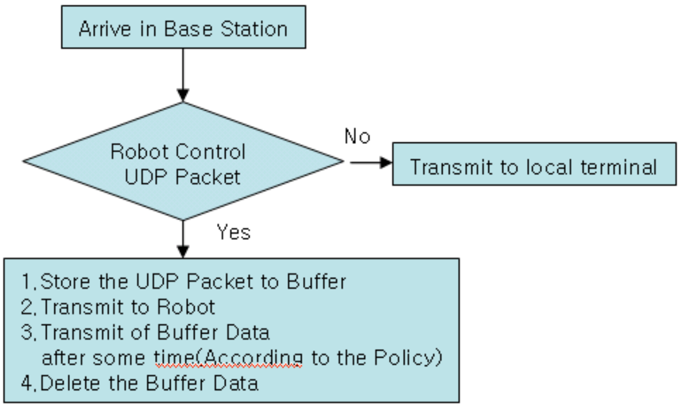
Flow chart of data processing at wired links



In wired links, packet loss is few and speedy transmission is possible. We select the UDP protocol for robot control data transmission. If router receives the UDP robot control packet (with PRI=15 in Ipv6), the router must immediately transmit the packet first of all. In wireless link base station, whether the received packet is robot control UDP or not, one copies the packet and transmit multiply or transmit the packet as it is.

FIGURE 3

Flow chart of data processing at Base Station of wireless links



In Fig.3, there might be many policies. If there happens a packet loss at the wirless link and there is a chance that the loss is noticed by the application, one can choice the interval time. For example, packet losses can be noticed by receiving ACK packets in application layer. Then the local retransmission interval time can be defined after the receiving the ACK packet.

I will comment the protocol which is about network consisting of mobile links between user and internet. In this case, the condition is more complicated than the links between Base Station and Robot. Detailed information will be provided at ITU presentation site.

Attachment 4 to Annex A

Source: Document 8F/645

1 Introduction

The essential technical and operational characteristics needed to support IP applications over mobile systems are as varied as the applications that may be supported over the Internet Protocol (IP). In the following sections, the basic capabilities required to support IP applications are illustrated and expanded upon to demonstrate how IMT-2000 CDMA multi-carrier supports various multi-media applications.

2 Basic Capabilities

2.1 Introduction

The Internet can be modeled as a collection of hosts interconnected via transmission and packet switching facilities. The most basic technical characteristic in the support of IP applications is the ability to convey data towards its intended recipient. A mobile system is typically involved in the final delivery of data to the intended recipient. This process may be further segmented into the encapsulations of IP packets for transport, and the ability to identify and route to the intended recipient.

2.2 Encapsulation

An IMT-2000 CDMA multi carrier system encapsulates the IP packet for transmission of the air interface using the general architecture defined in 3GPP2 C.S0001, which includes: the physical layer specified in 3GPP2 C.S0002; the MAC in 3GPP2 C.S0003; the LAC in 3GPP2 C.S0004; and upper layer signaling in 3GPP2 C.S0005.

The High Rate Packet Data Air Interface defined in 3GPP2 C.S0024 supports data rates up to 3.1 Mbps in the downlink and 1.8 Mbps in the uplink. Work is ongoing to support much higher peak data rates by combining up to 15 1.25 MHz channels.

2.3 Address Management

In IMT-2000 CDMA multi-carrier, IP addresses are assigned to mobile stations automatically using one of two mechanisms. The first mechanism uses the Point-to-Point Protocol, PPP, see for example IETF RFC 1661 and other references in 3GPP2 X.S0011. The second method uses the Dynamic Host Control Protocol, DHCP, see for example IETF RFC 2131 and other references in 3GPP2 X.S0011.

IP Addresses are typically invisible to subscribers via a mechanism that maps user-friendly names to IP addresses. IMT-2000 CDMA multi-carrier supports the Domain Name System, DNS, for name to address translation, see for example IETF RFC 1035 and other references in 3GPP2 X.S0011.

3 Quality of Service Considerations

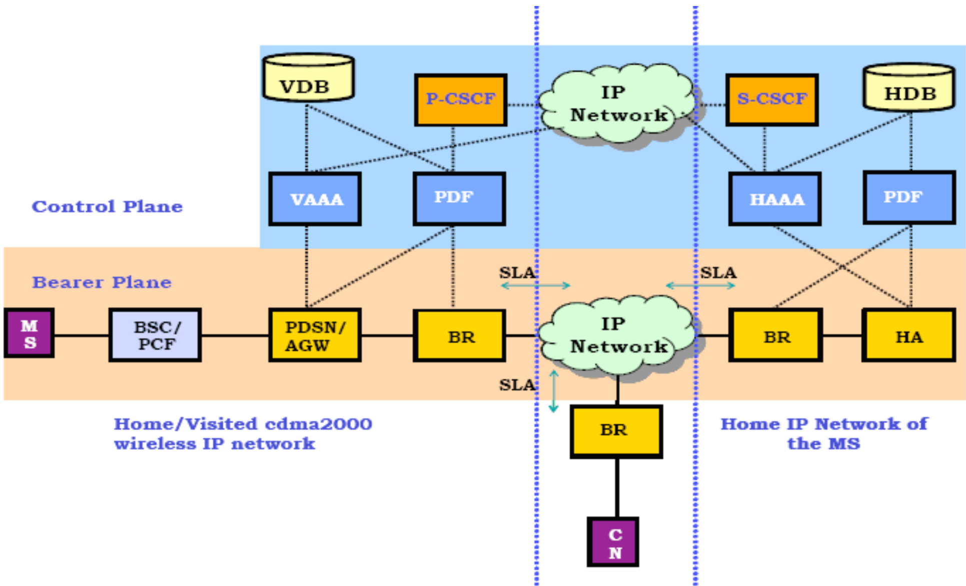
3.1 Introduction

The following information provides background information related to Quality of Service (QoS) considerations for an IMT-2000 CDMA multi-carrier wireless IP network and is derived from 3GPP2 S.R0079.

Requirements are defined to enable an IMT-2000 CDMA multi-carrier wireless IP network to provide End to End (E2E) QoS between a mobile station and a correspondent node. The E2E QoS network reference model involves several IP nodes. The two end points are the MS and the correspondent node (CN). The intervening networks span across an IMT-2000 CDMA multi-carrier wireless IP network that includes the radio link, the intermediate IP network, and the Edge IP network of the correspondent node. The E2E reference model can be viewed as a set of consecutive networks. The Figure below depicts an example of the E2E QoS architecture. (Note: the Figure below is for reference purposes only. It does not imply that all network elements shown are necessarily involved with E2E QoS.) E2E QoS may be provided by explicit management of QoS on the consecutive networks, or by provisioning, or a combination of both.

FIGURE: E2E

QoS reference model



The availability of E2E QoS functionality in the IMT-2000 CDMA multi-carrier wireless IP network provides the following benefits:

- E2E QoS would enable users to launch a variety of applications and experience their associated benefits in the wireless mobility context.
- E2E QoS would enable IMT-2000 CDMA multi-carrier wireless IP network providers to offer a variety of services and benefit from their associated revenue streams.

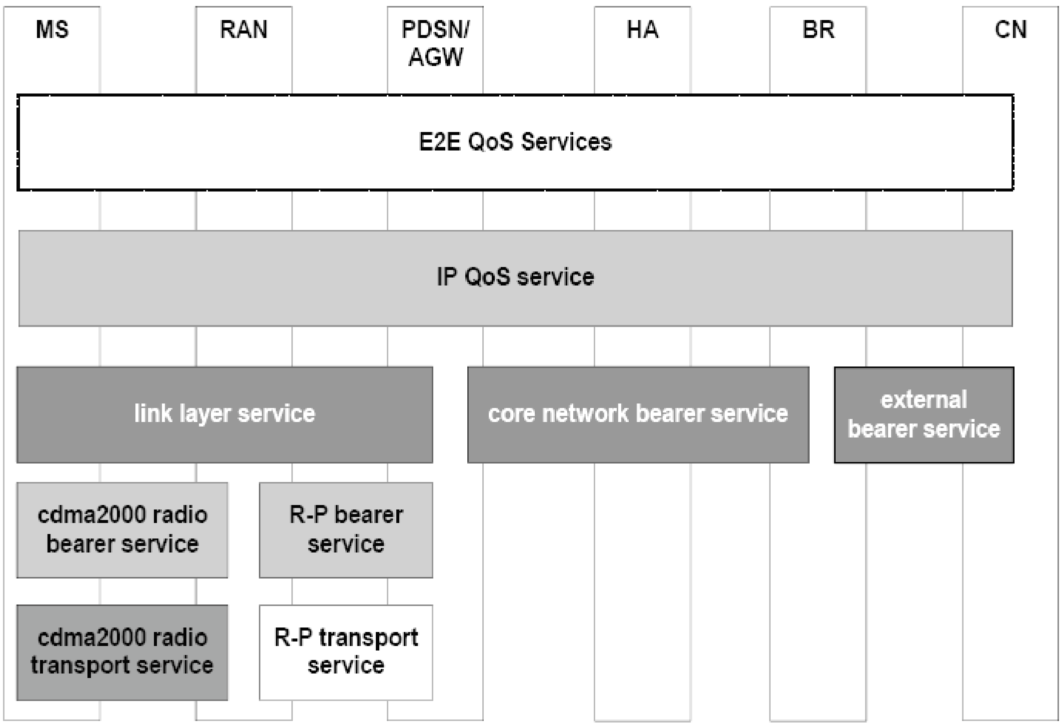
The following provides an example of an approach to E2E QoS in the IMT-2000 CDMA multi-carrier wireless IP network: The E2E QoS support in the IMT-2000 CDMA multi-carrier wireless IP network may be provided via one or more instances of a packet data service. The types of instances of a packet data service are identified as a main service instance or an auxiliary service instance. In the IMT-2000 CDMA multi-carrier wireless IP network, the radio resources should be allocated per service instance. In this context the purpose of a main service instance is used to provide resources in the IMT-2000 CDMA multi-carrier wireless IP network to meet the QoS requirements for the applications that may only require Best-Effort QoS support. However, to meet the QoS demands of applications that require better than Best Effort QoS, an auxiliary service instance can be used. The resource allocation for an auxiliary service instance is selective and is based on a characterization of QoS requirements associated with an application. One or more auxiliary service instances may be established by the MS based on the number of applications in use for an MS each requiring different QoS.

3.2 The IMT-2000 CDMA Multi-carrier E2E QoS bearer services

The E2E QoS support in the IMT-2000 CDMA multi-carrier wireless IP network attempts to reserve the necessary resources to ensure that the requested QoS requirements for a user's application are satisfied. If the necessary resources are not available in the IMT-2000 CDMA multi-carrier wireless IP network, an attempt should be made to negotiate a lower QoS. The following figure shows the different bearer services in a IMT-2000 CDMA multi-carrier network to satisfy subscriber's E2E QoS requests.

FIGURE: E2E

QoS architecture



E2E QoS Service: The application layer QoS between the end hosts identifies the QoS requirements, for example via SIP/SDP signaling protocol. The QoS requirements from application layer are mapped down to create a network layer session. The mobile terminal then establishes a link layer connection suitable for support of the network layer session. The QoS parameters received from the application layer are mapped to the corresponding IP layer signaling parameters as well as the link layer parameters.

IP QoS Service: In the E2E scenario, the mobile terminal can use the IP QoS service to control the QoS at the local and remote access networks, and diff-serv to control the IP QoS through the backbone IP network. Any IETF defined IP QoS signaling protocol can be used for different services. The entities that are supporting the IP QoS signaling should act according to the IETF specifications for int-serv and int-serv/diff-serv interworking. In addition to provision of E2E QoS Service and via IP QoS Service, QoS requirements may also be determined based on operator local policy/SLA.

Link Layer Service: The Link layer service currently does not provide any QoS capability. Support for QoS at the PPP layer (or any other link layer protocol that might be used in the future) is FFS.

External Bearer Service: The bearer services provided by the external network. e.g., the IP core network that is not owned and operated by the wireless service providers.

IMT-2000 CDMA multi-carrier Radio Bearer Service: IMT-2000 CDMA multi-carrier radio bearer services and their associated QoS parameters are defined in 3GPP2 C.S0017 and 3GPP2 C.S0024-A v1.0. This includes both the assured mode and non-assured mode QoS parameters. This service is enabled by the IMT-2000 CDMA multi-carrier radio transport service.

R-P Bearer service: The R-P bearer service is concerned with the QoS guarantee for the following service scenario: The bearer resources are allocated on the R-P interface in an attempt to meet the QoS requirements received from the mobile user as allowed by the network.

Core network bearer service: The core network in the IMT-2000 CDMA multi-carrier wireless IP network provides this type of bearer service between PDSN/AGW and BR.

IMT-2000 CDMA multi-carrier Radio Transport Service: This service is provided by the IMT-2000 CDMA multi-carrier physical layer that is categorized by the QoS classes and parameters based on the stringent requirements of the physical channels (FCH, DCCH, SCH, etc). Note that the MAC/Multiplex sublayer has to map the radio bearer QoS parameters (logical channel) onto the physical channel QoS parameters. The radio transport layer service is concerned with the physical radio channel payload data units produced and consumed by the IMT-2000 CDMA multi-carrier radio bearer service plus any signaling associated with those radio channels, e.g., common channel signaling, and call control messages and OAM. The radio transport service QoS should not be dependent on the definition of the radio bearer service QoS, or any higher-layer QoS definitions.

R-P Transport Service: The service provided by the R-P transport network to guarantee delivery of the R-P bearer services within their specified QoS limits.

4 Supporting Mobility

In 3GPP2 X.P0011-C, the requirements and procedures for Mobile IPv4 operation are specified based on a set of RFCs (including RFC 2002). The Mobile Station (MS) is able to use either a static Home Address or a dynamically assigned Home Address belonging to its Home Agent (HA) in the MS's home IMT-2000 CDMA multi-carrier network. The MS has a static HA address assigned regardless of whether the MS has a static or dynamic Home Address. The MS is able to maintain the Home Address persistent throughout the packet data session even when handing off between radio networks connected to separate PDSNs.

Mobile IPv4 operation is enhanced to support the scenario where the MS requests dynamic HA assignment in addition to dynamic Home Address assignment. A fast handoff in the context of an inter-PDSN handoff enables the MS's data traffic to traverse through the anchored PDSN even when the MS has moved to a radio network connected to a new serving PDSN. A tunnel is established between the anchored and serving PDSNs to transport the MS's data traffic. The fast handoff minimizes data loss when the MS is handing off between radio networks connected to separate PDSNs.

Work is ongoing towards 3GPP2 X.P0011-D, in which the requirements and procedures for Mobile IPv6 operation are specified based on a set of IETF RFCs (including RFC 3775). The MS is able to use either a static or dynamically Home Address and/or HA. The MS is able to maintain the Home Address persistent throughout the packet data session even when handing off between radio networks connected to separate PDSNs. Mobile IPv6 operates in two distinct operation modes: Bi-directional tunnel mode and Route Optimized mode. If bi-directional tunnel mode is used, all the MS's data traffic traverses through the MS's HA. If Route Optimized mode is used between the MS and a corresponding node, the data traffic exchanged between the two by-passes the HA and thus avoids the so-called triangular routing.

5 Security Considerations

5.1 Introduction

There are several security issues related to supporting IP applications whether the applications are run over a mobile system or not.

5.2 3GPP2 Security Features

Each 3GPP2 subscription has a User Identity Module (UIM), which can be either an integral part of the mobile or be removable. The UIM is a secure module that holds subscription information along with cryptographic keys, which are used to provide secure access to the network. For example, for MMD (see section 6), the keys in the UIM are used to provide mutual authentication between the user and network, when a user registers. Keying material derived from this authenticated is then used to integrity protect all further signaling traffic between the mobile and MMD and hence provide the mobile with secured access to services using MMD. The latest release of the specification (currently under development) will also provide confidentiality protection of this traffic.

3GPP2 is also developing a system called the Generic Bootstrapping Architecture. The major aim of this is to provide keying material to secure a variety of applications that run between the mobile and the network. The keying material is derived from the cryptographic keys that are already on the UIM.

6 Service Architecture

6.1 Introduction

The service architecture provides a secure, extensible framework under network operator control.

6.2 MMD overview and references

3GPP2 has adopted the IP Multimedia Subsystem (IMS) as the basis for the service architecture. 3GPP2's Multi-Media Domain (MMD) includes IMS and the IMT-2000 CDMA multi-carrier packet data network. The 3GPP2 MMD network provides third generation capabilities and is based on IETF protocols, including SIP, SDP, Diameter, and Mobile IP. It describes the system elements, interfaces, protocol specifications and procedures to provide the complete specification for the MMD core network.

6.3 MMD and regulatory features

MMD supports features that certain administrations may require, such as lawful surveillance of signaling and bearer traffic. MMD will also be extended to support VoIP with GPS-assisted position location for emergency services. IP applications over mobile systems should support the relevant specifications in this area.

7 Inter-working

7.1 Introduction

With the goal of providing a seamless user experience, IMT-2000 CDMA multi carrier extends support for IP applications to other wireless systems.

7.2 Wireless Local Area Networks

IMT-2000 CDMA multi carrier supports inter-working with Wireless LAN networks, with the intent of extending support for IP applications to the Wireless LAN environment while maintaining authentication and accounting aspects. The work has involved cooperation with IETF, and initial publication of 3GPP2 X.S0028 is pending completion of the RFC process in the IETF. Work is continuing to enhance the user experience with access to native services and seamless handoff.

7.3 GPRS

Support for IP applications is extended to a subscriber that has roamed into a GPRS network via the protocols and procedures defined in 3GPP2 X.S0034.

8 Spectral Efficiency

8.1 Introduction

The cost and limited availability of spectrum requires that the wireless technology transporting the IP traffic be as efficient as possible.

At the physical layer the measure of spectral efficiency is bits per second/Hz of spectrum used. Optimizing this requires a careful design of the system to minimize overhead and exploit the different QoS requirements of IP applications, the variability of the wireless channel, and the mobility of multiple users in the network. The IMT-2000 CDMA multi-carrier air interface is designed to do this by making use of multi-user diversity and intelligent schedulers.

Outlined in 3GPP2 C.S0024-A v1.0, IMT-2000 CDMA multi-carrier technology allows schedulers to quickly and efficiently favor users where the network is able to deliver more overall throughput while meeting the QoS requirements of users currently in less favorable radio conditions. Recognizing that IP applications require different levels of QoS, the IMT-2000 CDMA multi-carrier system design allows the scheduler to exploit this range of requirements and the variation in the wireless channel conditions among multiple users to maximize the delivery of throughput in the given spectrum. The system also allows the operator to dynamically balance throughput of a sector with delivered QoS requirements (e.g. throughput vs. latency).

Above the physical layer, the spectral efficiency for IP applications is maintained by using protocols with low overhead such as compression protocols. In particular, IP header compression protocols for IP multimedia such as RObust Header Compression, specified in IETF RFC 3095 and IMT-2000 CDMA multi-carrier-optimized header compression with zero-byte overhead detailed in 3GPP2 C.S0047-0 v1.0 are incorporated into the IMT-2000 CDMA multi-carrier system for this purpose.

9 Example Multi-media Applications

This section identifies a non-exhaustive list of IP applications that are supported by the IMT-2000 CDMA multi-carrier system standards.

9.1 Web-surfing

The QoS requirements for web-surfing generally require bursts of high-bandwidth while allowing for up to 1-2 seconds of latency. As further detailed in 3GPP2 C.S0024-A v1.0, IMT-2000 CDMA multi-carrier EV-DO system exploits the lax latency requirement of this kind of application to improve the overall throughput of the system. The scheduler is able to efficiently choose from among many users in diverse and varying radio conditions to favor those that maximize sector throughput while meeting the latency constraints for all web surfing users.

9.2 Voice-over-IP

The IMT-2000 CDMA multi-carrier system provides a range of service options and technologies for packetizing VoIP traffic that achieve the above advantages while carrying VoIP traffic over the air interface. The IMT-2000 CDMA multi-carrier 1x service options outlined in 3GPP2 C.S0063-0 v1.0 provide the same level of latency over the air interface as conventional circuit switched services while eliminating IP header overhead in the VoIP media traffic. The conventional IMT-2000 CDMA multi-carrier circuit-switched voice services already provide good capacity by supporting variable rate frames. The IMT-2000 CDMA multi-carrier 1x VoIP service option 61 provides this same capacity while supporting supplementary services carried over IP described in advantage 2 above.

The IMT-2000 CDMA multi-carrier EV-DO standard, 3GPP2 C.S0024-A v1.0, allows the network operator to trade off VoIP latency vs. sector throughput/voice capacity while sharing the same spectrum with other non-VoIP data applications. ROHC is also supported by the IMT-2000 multi-carrier system to reduce the IP overhead of the VoIP media traffic over IMT-2000 multi-carrier EV-DO.

The end-to-end system design for IMT-2000 CDMA multi-carrier VoIP services are described in 3GPP2 X.P0039-0 v1.0.

9.3 Streaming Audio and Video

The IMT-2000 CDMA multi-carrier standard 3GPP2 X.S0011-D provides procedures for the applications to reserve necessary bandwidth and jitter requirements with the radio network.

3GPP2 C.P0046-0 v1.0 describes how these multimedia streaming services are transported over IMT-2000 CDMA multi-carrier in a means that interoperates with general Internet streaming services.

9.4 Digital Video Telephony

The IMT-2000 CDMA multi-carrier EV-DO system is designed to allow an intelligent scheduler to provide the high bandwidth within the jitter and latency requirements while maximizing the sector throughput under such constraints. Since video telephony requires significant resources it is very important that the system be designed well for this application to minimize the impact on the system capacity and support a reasonable number of video telephony users.

The end-to-end system design for IMT-2000 CDMA multi-carrier Video Telephony services is defined in 3GPP2 X.P0039-0 v1.0.

10 Conclusion

In conclusion, significant support exists, and work is ongoing in 3GPP2 to enhance support of IP applications over mobile systems. This contribution has focused on the key high-level operational and technical characteristics required to support such IP applications over mobile systems including the basic capabilities necessary to support mobility over IP, the Quality of Service features which can be implemented, Service Architecture, Inter-working and the ability to support multimedia applications such as Voice-over-IP.

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3GPP2 X.S0013-005-0 v2.0,	IP Multimedia (IM) Subsystem Cx Interface; Signaling flows and message contents, 2005
3GPP2 X.S0013-006-0 v2.0,	Cx Interface based on the Diameter protocol; Protocol details, 2005
3GPP2 X.S0013-007-0 v2.0,	IP Multimedia Subsystem; Charging Architecture, 2005
3GPP2 X.S0013-008-0 v2.0,	IP Multimedia Subsystem; Accounting, Information Flows and Protocol, 2005
3GPP2 X.S0013-010-0 v2.0,	IP Multimedia Subsystem (IMS) Sh Interface signaling flows and message contents, 2005.
3GPP2 X.S0013-011-0 v2.0,	Sh interface based on the Diameter protocol, 2005
3GPP2 X.S0028,	cdma2000 Packet Data Services; Wireless Local Area Network (WLAN) Inter-working, 2004.
3GPP2 X.S0034,	cdma2000/GPRS Roaming, June 2005

Attachment 5 to Annex A

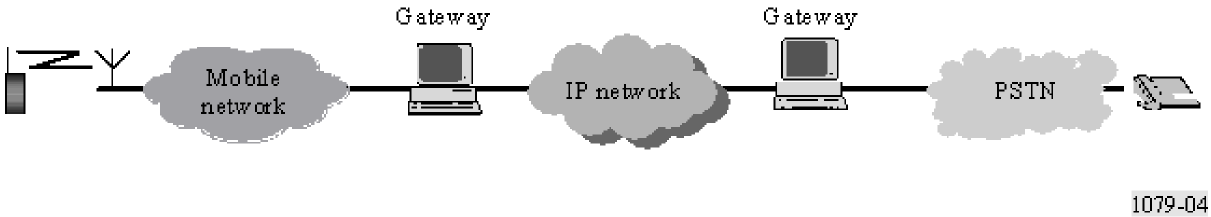
Source: Document 8F/609

The path of IP applications over mobile systems

There is a traditional relation between the existed PLMN, PSTN, and IP network, illustrated in Figure 1. The IP network mainly deals with data communications. That is from Recommendation ITU-R M.1079 (Performance and quality of service requirements for IMT-2000 access networks).

FIGURE 1

End-to-end system

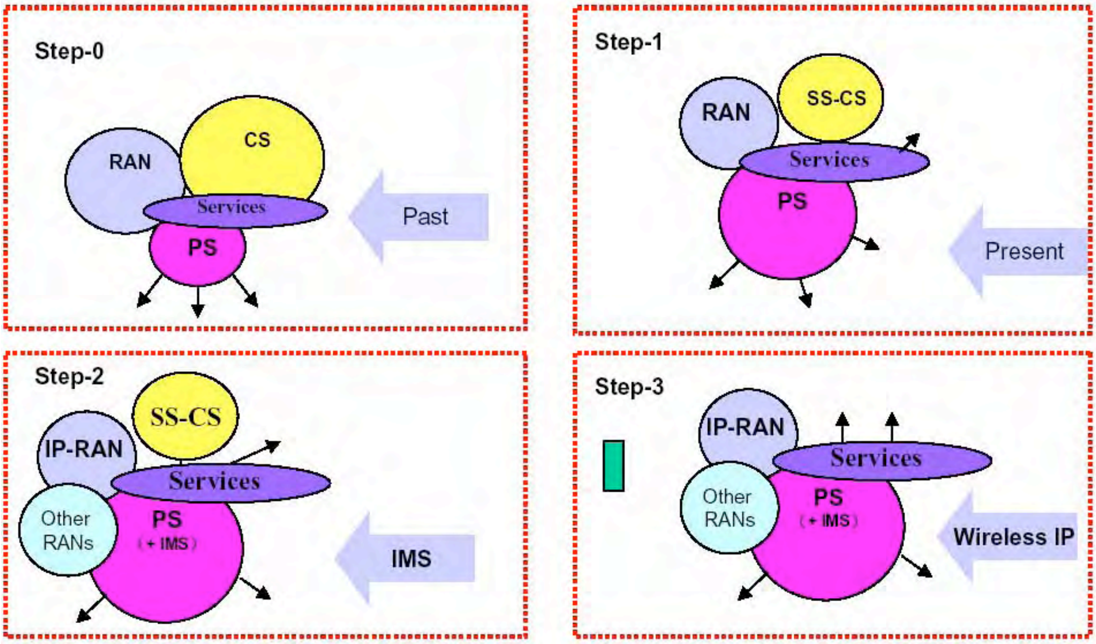


Modern digital technology allows different sectors, e.g. wireline telecom, data, wireless telecom, to be merged together. This convergence is happening on a global scale and is drastically changing the way in which both people and devices communicate. At the center of this process, forming the backbone and making convergence possible, are IP-based networks. IP-based systems offer significant advantages to operators and subscribers, from connectivity across a variety of devices, networks and protocols, to greater flexibility in the management, use and cost of network resources. IP is the guarantee of openness of the beyond 3G, which can merge data, voice and multimedia. All the components use IP protocols to provide transport for all types of bearer and signalling information in All-IP network. However, the transformation of the current mobile network to All-IP network architecture cannot happen overnight. IP network is enlarging into the mobile network and fixed network gradually.

For current mobile network, there are core network including Packet Switch (PS) domain, Circuit Switch (CS) domain, Service Deliver Platform (SDP) domain, and radio access network (RAN) domain. Up to now, PS domain and SDP domain have already applied IP technologies, as shown in Figure 2. It supports simultaneous use of the circuit-switched voice services and the packet-switched session services. Recently, IP is being utilized to both the carrier network and the signaling network in CS domain gradually, such as R4 in 3GPP and LMSD in 3GPP2. The key enhancements are Transcoder Free Operation (TrFO) and SIGTRAN. In general, there is no big problem if the special IP carrier network, e.g. CN2 of China Telecom, is deployed. The new bearer and signalling interfaces will gently be supported. An IP signalling network would replace the old SS7 telecommunications protocol, IP networks use some bandwidth-expensive mechanisms to achieve reliability.

FIGURE 2

IP Evolution in mobile networks



An individual user can be connected via a variety of different radio access systems to the networks. The interworking between these different access systems could be realized through a common IP-based core network with "optimally connected anywhere, anytime" manner. This IP-based core network shall be open to any service currently used and to be used in the future. The IMS (IP Multimedia Subsystems) based on SIP protocol of an IETF protocol is suggested to be access technology-agnostic so that the IMS may be implemented to not only IMT-2000 access technologies but also other IP access technologies.

– Non-IP-based systems (voice delivery) e.g. GSM, CDMA2000 and WCDMA {2GHz, WAN, Seamless, Voice/Data, Handset/Computer};

– IP-based systems (data services) e.g. 802.11 WiFi {2.4GHz, LAN, Hotpoint, Data/VoIP, Computer/Handset}, and 802.16 WiMax, 802.20 WBMA.

Consideration of IP RAN

An All-IP based B3G wireless network has intrinsic advantages over its predecessors. IP tolerates a variety of radio protocols, and lets you design a core network that gives you complete flexibility as to what the access network is. The core network provider can support many different access technologies, WCDMA, CDMA2000, Bluetooth, WiFi, WiMax, and some that we haven't even invented yet, such as some new CDMA protocols.

High rate access technologies, e.g. HSDPA/HSUPA and EV DO A/B, require the IP transmission over RAN. Combination of fixed and mobile access is called Connectivity access network (CAN).

IP wireless environment would further reduce costs for service providers by ushering in an era of real equipment interoperability. Wireless service providers would no longer be bound by single-system vendors of proprietary equipment. Future CAN will be integrated with all IP core network to interwork with other RANs including legacy RANs. Generally, 3G systems are regarded as non-IP based wireless access. But IP RAN emerges in R5 of 3GPP and EV DO of 3GPP2. SDOs are doing a study on access network re-design to support an IP based access network.

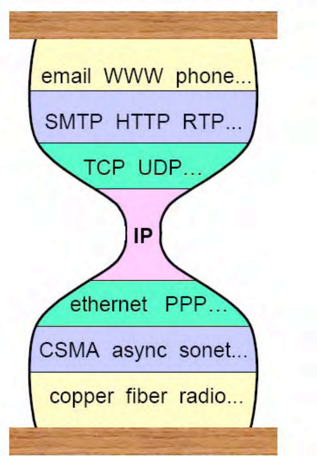
Generally, the RAN has the following features:

– IP RAN architecture becomes flat, all with IP bearing.

– Districted control makes some function reconfigure, e.g. the functions of RNC is weaken, its functions give to BTS.

– Maximize synergy among various transport infrastructure elements e.g., gateways, routers, etc.

– Intelligently splitting control and user planes to dynamically allocate capacity based on service demands.



vision

Figure 3 shows the vision of everything over IP and IP over everything. Everything over IP means IP network can bear various services. IP over every thing means IP may be transported over various media, such as fiber and radio.

IP over air interface

Dedicated allocation of physical resource such as circuit-switching transmission may take an advantage in terms of QoS guarantee but this causes inefficiency for low data rate or silent period of variable data rate services due to waste of radio resources. Then data for the services can be divided into IP packet, and each is transmitted through wireless medium.

Unfortunately, normal IP headers contribute a large overhead to the payload; for example, for VoIP, a packet has a total IP/UDP/RTP header size of 40 octets in IP v4 and 60 octets in IP v6. The size of the payload may be as low as 15-20 octets. Then the need to reduce header size for efficiency is obvious, especially for wireless links. Several methods (e.g. RFC2507/2508) have been proposed to reduce the header size. However, wireless links have characteristics that make header compression less efficient. They have to be robust enough to perform well in an environment with high bit error and packet loss rates. Robust header compression (ROHC) was proposed to conserve bandwidth in the narrow radio spectrum, reduce the packet loss rate over unreliable wireless link, and then to improve voice quality. This compression method was defined in the ongoing RFC3095, as part of the so-called packet data convergence protocol (PDCP) layer. All these tend to make IP spectrum efficient from the inefficient state.

For IP mobile network, there exist mobility and handover problems:

– Mobile IP supports macro mobility well. IETF proposes the Cellular IP to realize micro and pico mobility as the complement of the Mobile IP.

– A handover in any IP-based mobile network is a complex procedure. Typically, it takes quite a long time before the new access router gets the parameters describing the flow states associated with an incoming mobile node. trying to enable media independent seamless handover of a data session between 802.3/11/16, 3GPP, and 3GPP2. It is more difficult to attempt CS to VoIP seamless handover.Since mobile Internet will be always-on, operators realize the need for IP v6 addressing due to limited IP v4 address availability. In the future, both IP v4 and IP v6 users have to be accessed to applications and services on IP v4 networks.

Attachment 6 to Annex A

Source: Document 8F/437

Support of IP protocol with DECT

1 Introduction

This attachment describes the 2 basic ways in DECT to support IP services.

(A) Interwork the IP protocol in the fixed part as specified in TS 102 265, DECT access to IP based networks.

(B) Transparent transport of the IP packets across the air interface as specified in EN 301 649, DECT Packet Radio Service (DPRS)

The case (A) is described in section 2 and the case (B) is described in section 3 of this attachment.

2 DECT access to IP based networks

The TS 102 265 profile specifies the DECT interworking with IP networks. The profile specifies in particular DECT interworking with Session Initiation Protocol (SIP) and Mobile IP service. It is based and develops further the findings of the TR 102 010.

In regard to Mobile IP, IP addressing associated with the ‘Fixed Termination’ (FT) and/or with the ‘Portable Termination’ (PT) is specified. In regard to SIP interworking, Voice over IP (VoIP) and multimedia sessions are covered. In the case of voice, the VoIP is terminated in the FT and "normal" GAP based voice is used over the DECT air interface.

2.1 Reference configuration

Figure 1 describes the case when the IP protocol is terminated in the FP. This configuration may be preferable especially in the case of a DECT voice IP telephone (this does not exclude messaging or other data only services). Some issues that need consideration here in regard to providing interoperability between FPs and PPs are, e.g. unified way of transmitting the voice samples or other media streams to the PP and interaction with the signalling protocol use by the VoIP (e.g. SIP).

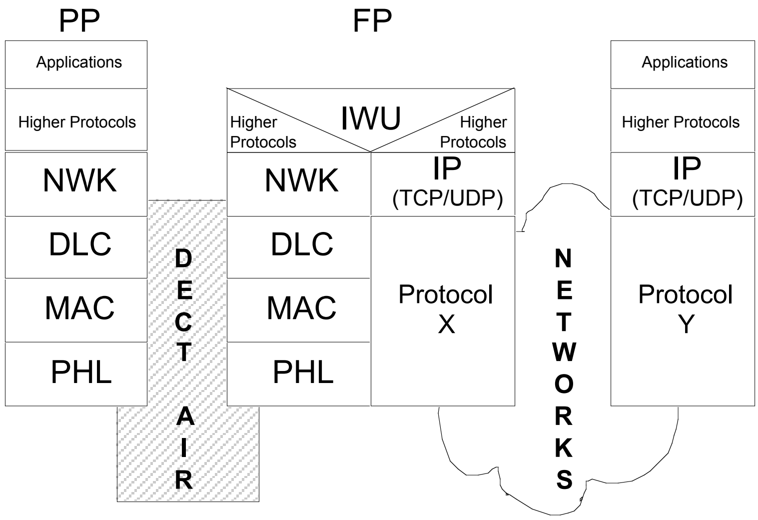


Figure 1: Reference configuration 2 (IP terminated at the FP)

2.2 IP-Roaming

The TS 102 265 specifies how the DECT procedures can be used to support roaming based on 'Mobile IP'. Figure 2 shows an example of a complete Portable Part IP roaming registration procedure.

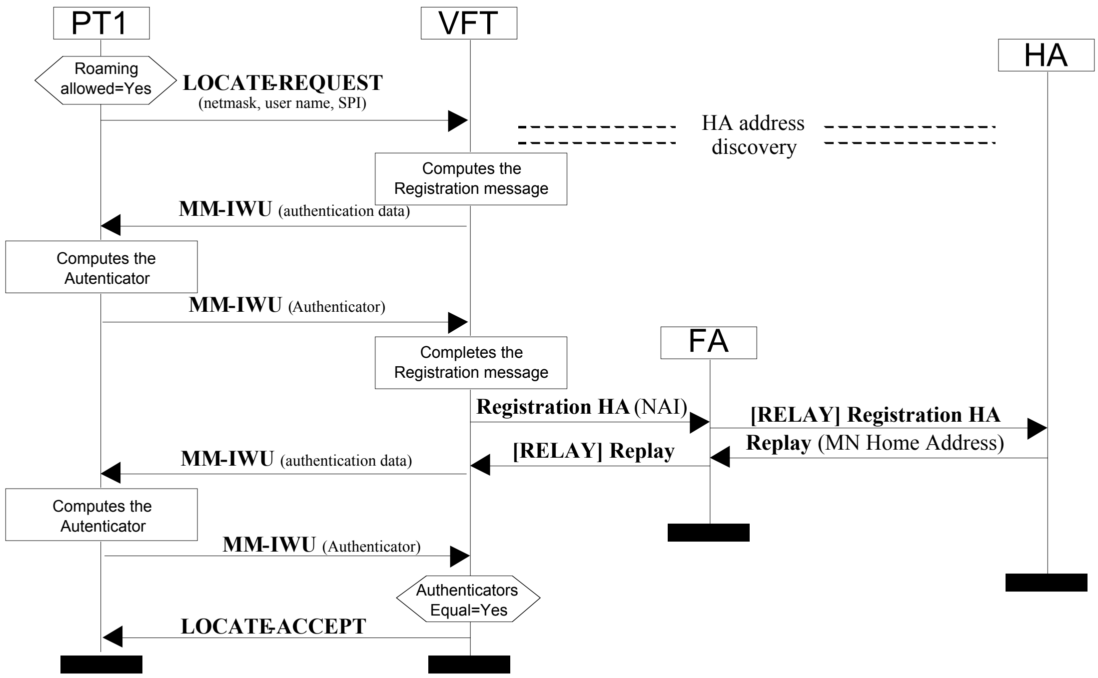


Figure 2: Successful PP mobile IP roaming registration (PT auth)

2.3 SIP Interworking

SIP is an application-layer signalling protocol that can establish, modify, and terminate interactive multimedia sessions over IP between intelligent terminals with one or more participants. These sessions include Internet telephone calls, multimedia distribution, and multimedia conferences. It is a clear text client/server protocol using Uniform Resource Locators (URL) for addressing (in this sense having a lot in common with HyperText Transfer Protocol (HTTP)).

SIP invitations used to create sessions carry session descriptions that allow participants to agree on a set of compatible media types (users may move between endpoints, they may be addressable by multiple names, and they may communicate in several different media - sometimes simultaneously). SIP makes use of elements called proxy servers to help route requests to the user's current location, authenticate and authorize users for services, implement provider call-routing policies, and provide features to users. SIP also provides a registration function that allows users to upload their current locations for use by proxy servers.

SIP runs on top of several different transport protocols enabling Internet endpoints called user agents (UA) to discover one another and to agree on a characterization of a session they would like to share. For locating prospective session participants, and for other functions, SIP enables the creation of an infrastructure of network hosts (called proxy servers) to which user agents can send registrations, invitations to sessions, and other requests.

A user agent represents an end system. In the context of the TS 102 265 a User Agent comprises a DECT Fixed Part and a DECT Portable Part and the UA activities may be provided either by the Fixed Part or by the Portable Part. As a Fixed Part can serve a number of Portable Parts, each tandem of the Fixed Part and a Portable Part may, but need not, represent an independent UA, i.e. the Fixed Part may be engaged in a number of different UAs, whereas a Portable Part may be engaged only in one UA at a time.

The Figure 3 gives an example how DECT can support a SIP session establishment and termination.

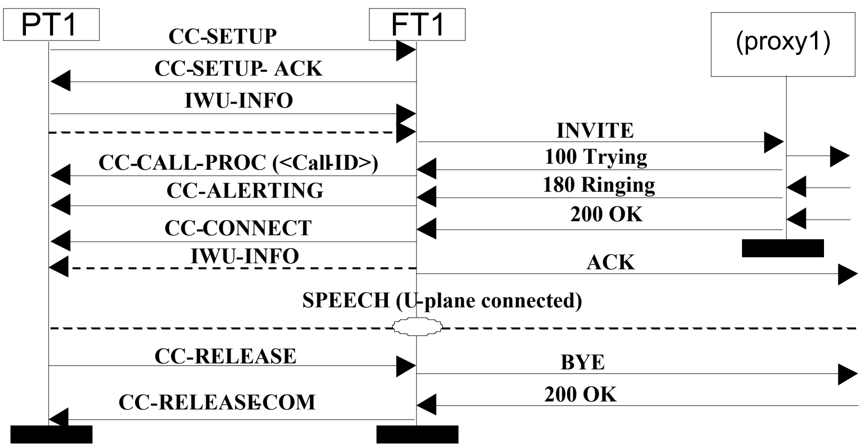


Figure 3: Successful SIP session establishment and termination Outgoing call (GAP)

3 DECT Packet Radio Service (DPRS)

The DECT Packet Radio Service, DPRS, EN 301 649 specifies common features and services for all packet data applications. This profile also serves as a base specification for other data profiles. DPRS does not contain GAP speech functionality but whenever needed (e.g. Call Control and Mobility Management procedures) it refers to the procedures defined in GAP, all additional procedure support necessary for data applications is explicitly specified in DPRS.

Interworking with V.24 interfaces, Ethernet, Token Ring LANs, direct interworking with Internet Protocol (IP) and PPP and, a Generic media encapsulation protocol allowing for various different media protocols to utilise one transport have been defined.

The standard contains specifications for applications for which a high degree of data integrity is necessary and includes connection oriented bearer services. A set of fast suspend and resume procedures is provided to overcome the drawbacks in regard to resource utilization that can be identified in most of the connection oriented service.

DPRS also extends the data stream service into environments, such as public services, where significant mobility is a characteristic. This service may be used to provide interworking with a voice-band modem service over public networks such as PSTN or ISDN.

Annexes to the DPRS specify a set of services that can be provided. There are two types of services:

- Frame Relay Service includes transport of protocols with user-delimited frames. DPRS defines the following frame-relay services:
    1) IEEE 802.3 (Ethernet);
    2) IEEE 802.5 (Token Ring);
    3) Internet Protocol (IP);
    4) Point to Point Protocol (PPP).
    5) Generic media encapsulation protocol
- Character Oriented service incorporates a packet assembling and disassembling (PAD) functionality to transport a stream data. DPRS incorporates the following Character Oriented services:
    - V.24 (asynchronous data).

USB interworking is provided as well.

3.1 Reference configuration (transparent IP transport)

Figure 4 shows the transparent transport of the IP protocol across the Fixed Part and the termination of the IP protocol in the PP.

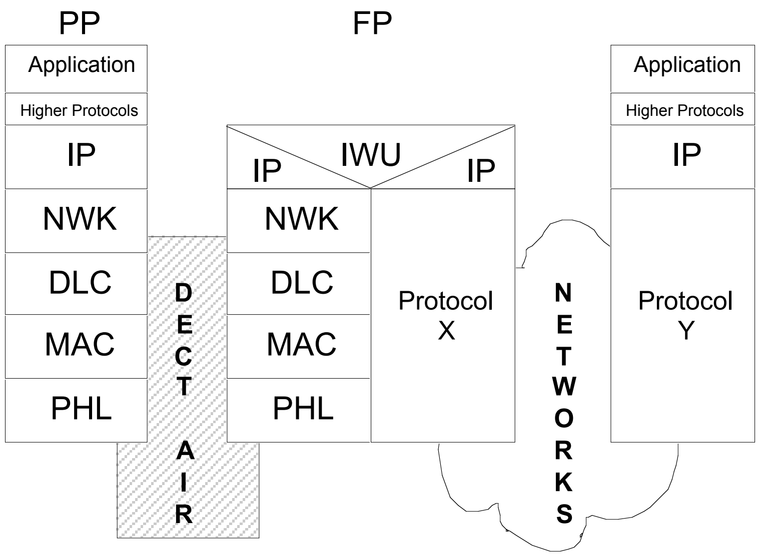


Figure 4: Reference configuration 2 (IP terminated at the PP)

3.2 Application Specific Access Profiles

Application Specific Access Profiles (ASAPs) identify a specific application scenario and select a subset of DPRS services for such applications.

The relationship between the standards and DPRS is shown in figure 5.

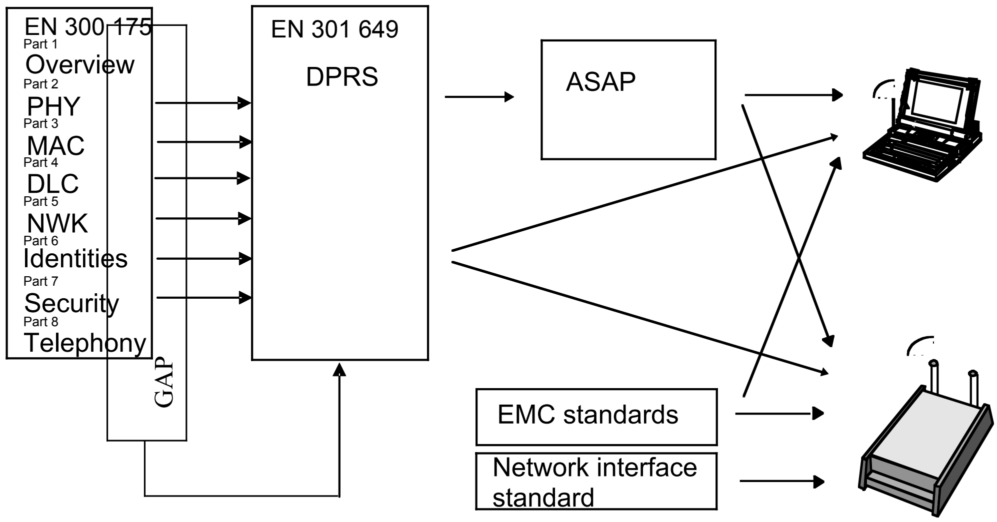


Figure 5: Standards relating to DPRS

The following 2 subsections describe ASAPs for Ethernet and V.24 Interworking.

3.2.1 Ethernet Interworking

TS 101 942 defines a data Application Specific Access Profile (ASAP) intended for enterprise, small office and home office (SOHO) and Home (residential/private) markets combining a selection of Ethernet Interworking DECT-DPRS data services. The TS 102 014 specify the Ethernet ASAP Test Specification (PTS) and the TS 102 013 (2 parts) specify the Ethernet ASAP requirement list and profile specific Implementation Conformance Statement (ICS) proforma respectively.

The aim of TS 101 942 is to guarantee a sufficient level of interoperability and to provide an easy route for development of DECT DATA LAN applications.

3.2.2 V.24 Interworking

TS 101 947 defines a data Application Specific Access Profile (ASAP) intended for enterprise, small office and home office (SOHO), and, home (residential/private) markets combining a selection of V.24 Interworking DECT-DPRS data services. The TS 102 012 specify the V.24 ASAP Test Specification (PTS) and the TS 102 011 (2 parts) specify the V.24 ASAP requirement list and profile specific Implementation Conformance Statement (ICS) proforma respectively.

The aim of TS 101 947 is to guarantee a sufficient level of interoperability and to provide an easy route for development of DECT DATA simple cable replacement applications.

4 Summary

This attachment gives a brief description how DECT can support IP applications and which DECT standards have been developed in this area.

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