Enterprize WEB and Multimedia Mobile Portal for Mobile Users

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Abstract. Users derive utility from services offered by the network. Lately, due to the multiplicity of WEB and multimedia applications as well as government deregulation, we have witnessed a corresponding increase in demand for a wide variety of mobile network services across a wide expanse, assembled together in an ad hoc manner. Images, voice, audio and video content will soon reach mobile terminals as part of mobile multimedia services paving the way to content-push services. On-demand mobile media services will be delivered to users via media streaming and download techniques that enrich mobile browsing and general content access. The paper describes the business opportunities, technology, and product strategies for the delivery of on-demand mobile WEB and media.

1 Introduction

An application hosting environment provides a solution that empowers mobile network service providers to deliver a vide variety of enhanced services over the wireless networks. Open service platforms are emerging with interoperability, programmability, scalability, and wide protocol compliance. In particular, new class of services will be presented to demonstrate the transformation of mobile services that is enabled through WEB presence. This article gives an overview how a new service portal can be created, listing all WEB and On-demand Media Services, could be deployed in a third party communication environment. The enterprize communication provides WEB access to presence information, instant messaging, and location. Some front runner services include converged data and wireless VPN and WEB/Internet based call center. These applications need good quality of service (QOS) provisioning. In the first case study the GPRS QOS provisioning will be presented. GPRS platform can be enlarged by WLAN networks, but both have to provide solutions for mobility, since mobile users are the most growing segments of the telecommunication market. In the second case study mobility analysis is carried out for GPRS and WLAN. In this paper no details are presented of voice delivery techniques like voice over IP (VoIP), this can be the focus of the future work. In the related work the ICEBERG architecture will be explained, which integrates data and voice, supports networks with diverse access technologies, and facilitates personal mobility among them.

2 Mobile WEB Service Technology

WEB service is an upcoming and promising middleware technology, which is based on common internet technologies like Extensible Markup Language (XML) [1], which is an interface definition language (IDL), defined by the Object Management Group (OMG), and the hypertext transfer protocol (HTTP) [2]. WEB services are strongly supported by major application platform vendors such as Hewlett-Packard, IBM, Microsoft and Sun Microsystem [3]. The 3rd Generation Partnership Project (3GPP), the European 3G standardization body, defines the Open service Access (OSA) [4] API as an open network technology independent interface, that makes network resources available to third parties. The 3rd Generation Partnership Project 2 (3GPP2), the North American and Asian 3G standardization body, is in progress of adopting this specification. Consequently service development will be possible that is suitable for both European and North American/Asian 3G networks. OSA has the following requirements [3, 4]:

1. To provide an alternative to CORBA for third party application development.
2. To use OSA architecture and standard would compromise ease-of-use, scalable with respect to the number of client applications and the number of interactions, or security.
3. To reuse existing OSA functionality.
4. To make the OSA interfaces easier to use for non telecommunication developers by abstracting OSA methods from the more complicated complexity.
5. To be suitable for deployment over the Internet. This includes such issues as firewall traversal and garbage collection of resources.

In general WEB services are classified for a set of standard-based techniques to deliver distributed network services [3] having the following characteristics [5, 6]:

1. In the so called stateless WEB architecture a HTTP listener responded to connection or disconnection requests to the server. This allows heterogeneous systems to work together.
2. In the so called loosely coupled service oriented architectures, the service advertises its contact information and interfaces in a directory service. Clients query the registry for services and retrieve contact information. The implementation details, such as the operating system or the programming languages, are hidden from the clients.
3. XML-based messaging is used between systems that use different component models, operating systems, and programming languages. This means, XML-encoded messages are sent and received.
4. WEB service technology does not standardize an execution platform, (traditionally distributed components were used) but only the interoperability requirements [7]. So long WEB services are interoperability and platform independent.

2.1 Web Service Stack

The Web Service Stack consist of a set of Internet standards.

1. The most important XML based standard is the simple object access protocol (SOAP) [8]. SOAP specifies a mechanism to do remote method invocation by exchanging XML, HTTP or SMTP messages between WEB client and WEB services.
2. The second major standard is the WEB service description language (WSDL) [9]. WSDL describes in XML format the WEB service operations, protocol bindings, and protocol message formats and contains information about the service endpoint, which is the address where the service is deployed.
3. The last standard is the universal description, discovery, and integration (UDDI) [10], which stores the WDDI definitions and provides a way to locate WEB services based on their specific characteristics.

3 On-demand Mobile

Multimedia Technology The basic technology of the wireless Internet is similar to its fixed Internet counterpart, but the fundamental challenges are not, and the associated services will probably also differ.

3.1 Mobile Multimedia Fundamentals

Mobile Messaging

1. Short message service (SMS) is the basic.
2. Enhanced messaging services (EMS) are media-rich.
3. Multimedia messaging service (MMS) is well on the way.

Mobile WEB Technology In case of the wireless application protocol (WAP) [11] and access to the WEB content, two important enhancements were developed by general packet radio service (GPRS) and wideband code-division multiple access (WCDMA) services:

1. Media download
2. Media streaming.

In GPRS and WCDMA mobile networks, the data transmission rates has been increased from 40 to 384 kbit/s for typical wide area coverage.

Synchronized media integration language (SMILE) HTMP and wireless markup language (WML) [12] are limited in their abilities to deal with continuous media types, such as audio and video. SMILE [13] has been developed, therefore, to express relationships, in space and and time, between media elements.

3.2 Content Distribution

The distribution of mobile media content has tree phases:

1. The content creation process starts with content production. Live events are captured by cameras, VCRs, audio sources and CDs. If the raw content exists only in analog format, it must first be digitized. results in distinct media files. Row digital content needs editing and coding.
2. The distribution of content to application and media servers in the network from which the content can be accessed on-demand. This means: content aggregation, storage and management of specific content and delivery to the end-user. Before delivery, the content must be transformed into a format that is compatible with the end-user device (physical display size, processor power) and the network (protocols and available bit rates). The process of delivering content contains two options: media file download and media streaming.
3. Content presentation of media client in the terminal. This means: media recaption, decoding, presentation and layout.Optional playback and local storage can be available.

To transport video content, for example, over low bit rate (64 kbit/s), the video resolution must first be decreased, typically using quarter CIF (common interchange format) at 176 x 144 pixels and 10 to 15 frames per second. Moreover, the video must be compressed to further reduce bandwidth. Greater motion and more complex textures require larger bandwidth. Sporting events demand more than 256 kbit/s bandwidth and their storage requirement is more than 1000 kbytes computed for clip duration of 30 seconds., whereas news services demand less bandwidth like 16 kbit/s with storage requirement of 40 kbytes/30s.

3.3 3GPP Media Protocols

The 3GPP has defined the TCP/IP over GPRS bearer module for protocols used for on-demand mobile media. Several layers can be grouped into GPRS and TCP/IP protocols:

1. Presentation layer. This layer manages the user interface and the rendering of the features provided by the lower service layer. Mostly it is composed of HTML or Java servlet pages (JSP) pages. This layer is also responsible for handling user interactions and for updating presence information on the user. JAIN service logic execution environment (SLEE) [14] can provide a set of API interfaces to support application execution, like enabling dynamic execution of HTMP pages.
2. Session layer. This layer provides general services such as logging, database connectors, administration, intercomponent messaging services, and user data representation. These components are shared by the applications, and they are the basic blocks to build converged services. SLEE can provide an infrastructure to support interprocess communication through its message handling facilities. Depending on the service, several protocols can be used, like RTSP.

3. Communication layer. This layer interworks and manages the various network resources, spans multiple telecommunication boundaries: telephony, presence capability, location based services, and instant messaging through the presence and availability management (PAM) [15] services. On this transport layer either UDP or TCP is used.
4. Network layer. In this layer internet protocol (IP) interconnects all network elements between destination and source node.
5. Other supporting functional layers are: databases, measurement and billing.

Hypertext transport protocol (HTTP) Provides the semantics of requesting and transferring information including media objects between servers and clients in a distributed and collaborative way. HTTP is used on the top of TCP.

Real-time transport protocol (RTP) RTP is a protocol for real time data [16, 17]. Runs on the top of UDP. Used in IP telephony or in sending audio or video data over packet networks. RTP adds a time stamp and a sequence number to each UDP packet in a special RTP header. Time stamp is used for playing back the media at the correct speed. The sequence number is used by the recipient to detect the loss of packets.

Real time streaming protocol (RTSP) RTSP is used as a session control protocol for media streaming applications [18]. If a client streaming application establishes a session with media streaming server, the following can be asked:

1. Start streaming media.
2. Pause, back-up and relay, and fast forward streaming media.
3. Stop streaming and disconnect the session.

RTSP is usually used on the top of TCP, but can be used on the top of UDP as well.

4 Case Study One: General Packet Radio Service (GPRS)

GPRS offers efficient end-to-end wireless packet data services in GSM network. Despite the effort of the standardization bodies to refine quality of service (QOS) handling, taking into account the needs for application and subscriber differentiation, today's GPRS systems are generally considered as ”best effort” with its packet data protocol (PDP), while in release 1999, the traffic flow template (TFT) was introduced both to universal mobil telecommunication system (UMTS) and to GPRS release 99 [19].

4.1 Packet Data Protocol (PDP)

The PDP context is created by activation signalling, while a logical connection is made between the mobile station (MS) and the gateway GPRS support node (GGSN), and an IP (PDP) address is allocated to the MS. Once address allocation is done, the MS can receive data from external networks. The QOS, along with other parameters, is negotiated, and it can be changed (renegotiated) during the connection. This means: network elements are able to reduce the QOS, but the MS can renegotiate the QOS level up and down. UMTS as well as GPRS release 1999 has enabled to use of several PDP contexts (each having different QOS) per PDP addresses.

4.2 Traffic Flow Template (TFT)

While the number of simultaneous PDP contexts per user has grown, a mechanism to select an appropriate PDP context (QOS) had to be developed. Each TFT contains one or more filters that are signalled to the GGSN during PDP context activation or modification.Each PDP context has one TFT, but each TFT might contain up to eight filters. By means of the MS initiated PDP context modification procedure, any TFT can be modified. A PDP context can never have more than one TFT associated with it. A given TFT has a unique evaluation procedure index for one PDP address, and at least one of the following attributes:

1. Source address and subnet mask,
2. Protocol number (Ipv4)/ next header (Ipv6),
3. Destination port range,
4. Source port range,
5. IPsec security parameter index (SPI),
6. Type of service (TOS) (Ipv4)/ traffic class (Ipv6) and mask, and
7. Flow label (Ipv6).

During the PDP context activation, a filter evaluation priority is attached to each filter. The filter parameters are compared to the received packet. The subsequent filter parameters are evaluated until a match is found. If match is found, the evaluation is aborted and the packet is forwarded to the MS for downlink traffic on the associated PDP context.

5 Case Study Two: Mobil Internet Protocol (MIP)

MIP provides IP-based layer 3 mobility management between air interface technologies (and between code division multiple access (CDMA) packet data switching nodes [PDSNs]). A mobile device can seamlessly change its internet point of attachment while maintaining its logical link and network connectivity using a static or dynamically assigned IP address that it retains throughout the data session. MIP also provides strong authentication techniques. Thus, the MIP is a set of protocols [20] is well suited to provide mobility and mobile VPNs (MVPN) connectivity. MIP defines three network components:

1. The foreign agent (FA). FA is a router supporting MIP protocol in the foreign (visited) network. The FA function is located in the PDSN for CDMA, in the gateway GPRS support node (GGSN), and in the edge router or WLAN gateway for WLAN. The FA provides a care-of-address (COA) for MN.

2. The home agent (HA). HA is a router supporting MIP protocol in the user's home network. The HA function is located in the PDSN for CDMA, in the GGSN, and in the edge router for a third party data center. The HA maintains the MN's current COA information and tunnel datagrams into the MN's COA, when MN is away from home.
3. The mobile node (MN).

3G packet data technologies are designed to support mobility and are well suited to provide ubiquitous high speed wireless data service. However, data rates may fluctuate widely due to changing radio frequency conditions.

5.1 WLAN Access

The WLAN is complementary to 3G network. The WLAN only shares the 3G back office's AAA infrastructure, avoiding mixing traffic from the two types of access technologies. This keeps the two radio networks completely separate and independent of one another and requires no changes to IEEE 802.11 nodes. IEEE 802.11 technology is designed to provide wireless access to stationary or low mobility (pedestrian speed) users. IEEE 802.11b offers sustained data rates an order of magnitude higher than 3G but has a more limited range. WLAN is fundamentally a layer 2 bridging system and lacks of formal mobility management found in 3G systems.Hence, it is best suited for hot spots and in-building use. Mobility is only between access points residing on the same subnets, using the concept of MAC address reassociation with a different access point residing on the same subnet. Internetwork roaming and mobility are outside the scope of IEEE 802.11, and can only be achieved by relying on higher layer protocols.

6 Related Work: ICEBERG Architecture

ICEBERG architecture [21], aiming at operation in a wide area, requires deployment of ICEBERG points of presence, creation of provider/administrative domains. It also require service level agreements among the providers. It introduces an ad hoc signaling protocol, and requires an ICEBERG unique ID. The approach is revolutionary. Any existing wireless/PSTN network beyond its access system is replaced with an ICEBERG network plan. It has a super distributed home locator register (Super-DHLR), which has well-defined interactions with popular signalling protocols, and does not intervene in session/conference control or network and QOS management.

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8 Conclusion

First, we develop workload models to describe various IP-based applications with varying degree of sensitivity to latency and data loss rate [22]. We consider TCP connections and WEB traffic as background traffic. We study the design of QOS mechanisms both at the network and application layer to deliver in a scalable and bandwidth efficient manner, while providing graceful degradation under heavy network load. In the future we examine how to deploy feedback mechanisms in transport protocol like RTP and RTSP to address application interactive requirements.

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