Effective VoIP Call Routing in WLAN and Cellular Integration

Yi-Bing Lin, *Fellow, IEEE,* Whai-En Chen, *Member, IEEE,* and Chai-Hien Gan, *Member, IEEE*

***Abstract*— In cellular-WLAN integration, a dual-mode mobile station (MS) typically disables the WLAN module for power saving. A major problem is that for an incoming VoIP call (or data session), the MS will not be able to receive this call from the WLAN. It turns out that the call is directed to the cellular network. This letter proposes a simple push solution where an MS can accurately detect a VoIP call from paging signaling of the cellular network. Then the WLAN module of the MS is turned on and the VoIP call is connected to the MS through the relatively inexpensive WLAN.**

***Index Terms*— Call control, mobile network, push, SIP, UMTS, VoIP, WLAN.**

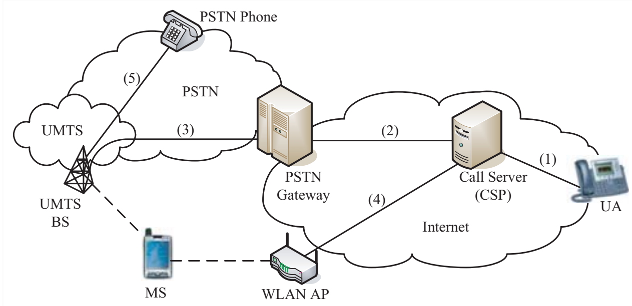

Fig. 1. WLAN and cellular integration environment.

I. Introduction

In RECENT years, many studies have been conducted to investigate *Wireless LAN* (WLAN) and cellular (such as *Universal Mobile Telecommunications System* or UMTS) interworking that extends mobile services to the WLAN environment. In this interworking, a dual-mode *Mobile Station* (MS) equipped with both WLAN and cellular modules is utilized, and the WLAN serves as an access technology to the cellular system, which scales up the coverage of mobile services.

Fig. 1 illustrates a WLAN and cellular integration environment where the MS typically attempts to access the WLAN first for lower costs and higher bandwidth connection. If the WLAN is not available, the MS then tries to access the cellular network. Due to large power consumption of the WLAN module, the MS typically turns on the cellular module and turns off the WLAN module. The WLAN module is turned on only when the user attempts to access the WLAN. A major problem of this usage style is that, for example, the MS cannot receive the incoming *Voice over IP* (VoIP) calls from the WLAN when the WLAN module is off.

In [2], [3], we proposed a push mechanism for VoIP incoming calls to WLAN-UMTS dual mode MSs. This approach utilizes *Short Message Service* (SMS) to alert the MS. In [4], the SMS mechanism is replaced by the normal cellular paging. Whenever the MS receives an incoming cellular call, it always attempts to connect to the WLAN first. If successes, the call is connected through the WLAN as a VoIP call. If fails, the MS accepts the cellular call. One problem about this approach is that the MS cannot distinguish a normal cellular call (path (5) in Fig. 1) from a VoIP call that is setup from the IP network (path (1)$\rightarrow$ (2) $\rightarrow$ (3)). Therefore, for a normal cellular call, the call setup is delayed because the MS always attempts to connect to the WLAN first. The MS always experiences the WLAN connection failure before connecting to the cellular network.

To resolve this problem, we propose a simple approach where the MS can detect the "originating network" of the calling party. In a typical Internet and *Public Switched Telephone Network* (PSTN) interworking example illustrated in Fig. 1, a *Session Initiation Protocol* (SIP) *User Agent* (UA) is connected to a SIP Call Server. The Call Server then sets up the call to the PSTN through a PSTN Gateway [1]. In a typical PSTN exercise, a PSTN Gateway is assigned an SS7 number (say, `936012xxx`), or a group of leased lines whose telephone numbers have the same prefix (e.g., `936012`). In our approach, the push mechanism is implemented in the Call Server as described in [2], [3]. The resulting network node is called *Call Server with Push* (CSP). The CSP maintains a *Dual-mode MS* (DM) *Table*. This table is used to track the call setup status of dual-mode MSs. The MS maintains a *PSTN Gateway Table*. For every PSTN Gateway connected to the CSP, the SS7 number of the PSTN Gateway (e.g., `936012xxx`) and the corresponding fully qualified domain name of the CSP (e.g., `nctu.com`) are stored in an entry of the table. In the next section, we show how the DM table and the PSTN Gateway table can be used to correctly activate an MS with turn-off WLAN module to receive an incoming VoIP call.

Manuscript received April 18, 2005. The associate editor coordinating the review of this letter and approving it for publication was Prof. Yannis (Ioannis) Psaromiligkos.

The authors are with the Dept. of Computer Science and Information Engineering, National Chiao Tung University, Taiwan (e-mail: {liny, wechen, chgan}@csie.nctu.edu.tw).

Digital Object Identifier 10.1109/LCOMM.2005.10027.

II. The Call Server with Push (CSP) Approach

Consider the SIP call setup procedure from an IP host (a UA) in the external data network (e.g., Internet) to an MS. The UMTS phone number of the MS is `+886936105401` and the

1089-7798/05$20.00

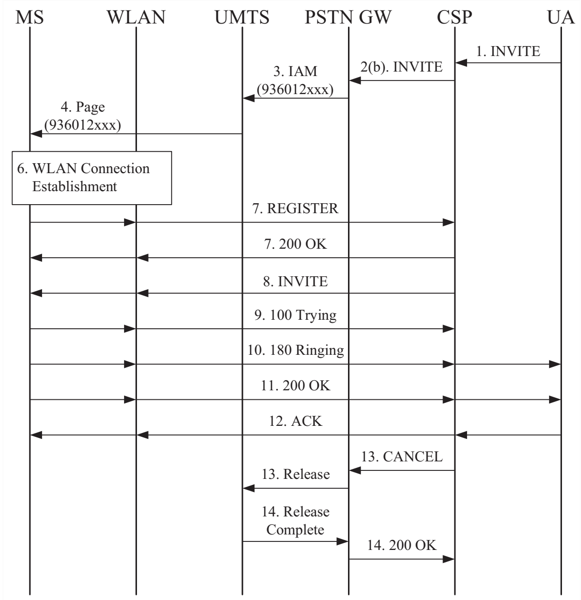

Fig. 2. Message flow for an incoming call to the dual-mode MS.

fully qualified domain name of the CSP is `nctu.com`. Fig. 2 illustrates the message flow and is described as follows.

Step 1. To set up a call to the MS, the UA sends an INVITE message to the CSP (Fig. 1 (1)). The INVITE message contains the SIP *Universal Resource Identifier* (URI) of the MS, i.e.,

`+886936105401@nctu.com`

and the *Session Description Protocol* (SDP) that describes the *Real-Time Transport Protocol* (RTP) information of the MS. The RTP information includes the IP address and port number of the UA.

Step 2. To resolve the SIP URI in the INVITE message, the proxy function of the CSP identifies the contact information of the MS.

Step 2 (a). If the MS has registered to the CSP, the CSP forwards the INVITE message to the MS directly, and follows the normal SIP call setup procedure to connect the call.

Step 2 (b). If not, the CSP routes the call to the PSTN according to the phone number `+886936105401`. Specifically, the CSP forwards the INVITE message to the PSTN Gateway (Fig. 1 (2)). The CSP also creates a record (i.e., a DM record) in its DM table for the MS to indicate that the call is set up to the PSTN.

Step 3. The PSTN Gateway generates an SS7 Initial Address Message (IAM) containing the caller ID `936012xxx` (the SS7 number of the PSTN Gateway). Since the destination number `+886936105401` is a UMTS MS *Integrated Services Digital Network* (ISDN) number, the IAM is sent to the UMTS network (Fig. 1 (3)).

Step 4. Finally, the UMTS *Base Station* (BS) pages the MS. The message carries the caller ID `936012xxx`.

Step 5. The MS checks its PSTN Gateway table to see if `936012xxx` is found.

Step 5 (a). If so, the corresponding fully qualified domain name of the CSP (i.e., `nctu.com`) is retrieved. Step 6 is executed.

Step 5 (b). If not, the MS answers the paging signal from the UMTS BS, and follows the normal UMTS procedure to connect the call.

At Step 5, through the caller ID, the MS accurately detects if the incoming call signaling comes from path (5) (Step 5 (b)) or from path (1)→ (2) → (3) (Step 5 (a)).

Step 6. The MS attempts to access the nearby WLAN network. If successes, Step 7 is executed. If no WLAN access is available, Step 5 (b) is executed.

Step 7. The MS registers its contact address to the CSP (the registrar function) through a SIP REGISTER message (Fig. 1 (4)). The registrar function updates the MS's contact address information, and returns a 200 OK message to the MS. Since the DM record indicates that the call is being set up for the MS (see Step 2 (b)), the CSP determines that the call should be re-connected to the MS through the WLAN.

Step 8. The CSP (the proxy function) forwards the INVITE message to the MS.

Step 9. Upon receipt of the INVITE message, the MS replies a 100 Trying message to indicate that the call is in progress.

Step 10. The MS plays an audio ringing tone to alarm the called user and sends a 180 Ringing message to the UA through the CSP. Upon receipt of the 180 Ringing message, the UA plays an audio ringback tone to the calling user.

Step 11. When the called user picks up the handset, the MS sends the final 200 OK message to the UA. The 200 OK message includes the SDP that describes the RTP information of the MS.

Step 12. Upon receipt of the 200 OK message, the UA sends an ACK message to acknowledge the MS. The call is connected.

Step 13. After Step 12, the CSP cancels the call setup procedure to the PSTN Gateway by sending a CANCEL message. The PSTN Gateway sends an SS7 Release message to the UMTS network.

Step 14. The UMTS replies an SS7 Release Complete message to the PSTN Gateway. The PSTN Gateway replies a 200 OK message to the CSP to indicate that the call is successfully canceled.

At this point, the voice path is (1)↔(4). We note the following:

- If any one of Steps 7-12 fails, the MS will execute Step 5 (b) to accept the cellular call (not shown in the call flow).
- For a non-VoIP incoming call from UMTS, the MS accepts the call immediately (Step 5 (b)). Therefore the extra delay in [4] is avoided.
- If the MS has already registered at the CSP, then the call is set up through the WLAN directly at Step 2 (a).
- If the MS cannot connect to any WLAN access point, it accepts the cellular call at Step 6.
- There may be several CSP-PSTN Gateway pairs in the MS's PSTN Gateway table, and the MS can detect the VoIP calls from various PSTN Gateways.

III. SUMMARY

In a WLAN and cellular integration network, a dual-mode MS typically disables the WLAN module to reduce the power consumption. Therefore, the incoming VoIP calls to the MS cannot be connected through the WLAN. To address this issue, we proposed a *Caller Server with Push* (CSP) mechanism where the MS can accurately detect the VoIP calls through cellular paging, and then the CSP can re-route the call through the WLAN. In our approach, the SIP server is slightly modified to re-direct call setup. The MS is slightly modified to implement a PSTN Gateway table. For the demonstration purpose, we used SIP VoIP over WLAN-UMTS integration as an example. Our solution can be easily extended for cellular networks such as GSM, GPRS, or cdma2000, and WLAN can be replaced by WiMAX. The CSP approach is pending ROC and USA patents.

ACKNOWLEDGEMENT

This work was sponsored in part by NSC Excellence project NSC93-2752-E-009-005-PAE, NSC 93-2213-E-009-100, and NTP VoIP Project under grant number NSC 94-2219-E-009-002. IIS/Academia Sinica, and ITRI/NCTU Joint Research Center.

REFERENCES

[1] Collins, D. *Carrier Grade Voice Over IP*. McGraw-Hill, Singapore, 2001.
[2] Lin, Y.-B., Lo, Y.-C., and Rao, C.-H. A Push Mechanism for GPRS Supporting Private IP Addresses. *IEEE Communications Letters*, 5(1), 2003.
[3] Feng, V. W.-S., Wu., L.-Y., Lin, Y.-B., and Chen, W.-E. WGSN: WLAN-based GPRS Environment Support Node with Push Mechanism. *The Computer Journal*, 47(4):405–417, 2004.
[4] Tsao,S.-L., and Cheng, E.-C. PIANO: A Power Saving Strategy for Cellular/VoWLAN Dual Mode Mobiles. submitted for publication, 2005.