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# Development of an intelligent VoWLAN services for patient care improvement

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## ABSTRACT

**Since the rapid exchange of information and collaboration with colleagues are indispensable for the quality of care. Effective communication between care givers has been recognized as a critical factor on the high quality of patient care. Communication in medical environment is often intense and time critical, including laboratory result, complex consultation and advice, which require high degree coordination among care professionals. Nevertheless, there are some deficiencies existed in the actual state of the communication system in hospitals, such as waiting time for call back and inducing interruptions. Current technological solutions should allow developing a novel intelligent communication system. In this paper, the proposed HPC2S (Hospital Patient Care Call System) services based on VoWLAN (VoIP over WLAN) will achieve the needs of timely and efficient communication among care professionals.**

**Keywords:** Intelligent Communication System, Voice over IP, Wireless LAN

## 1. INTRODUCTION

Due to the complexity of patient care and information access the requirements of related communication between patients and care professionals are augmenting. It has been demonstrated that effective communication among members of care team is essential in the delivery of high quality of patient care [1]. Without the waiting times on hold and the traveling back and forth to a phone, a significant timesaving directly translates into improved patient care. Obviously, an intelligent, cost-effective communication system is important to improve the quality of service. Some wireless communication systems provide effective communication, such as Personal Handy-phone System (PHS). However, it cannot rapidly deploy new service model, such as one number service, 3-way communication service, etc.

IEEE 802.11 based wireless LANs (WLANs) have grown remarkably and spread rapidly in diverse settings [2]. The market for WLANs has demonstrated a tremendous growth in recent years [3]. It is evident that there was a rapid increasing popularity of 802.11 WLAN hotspots deployed in residence buildings, enterprises, and public areas [4]. There are miscellaneous applications to WLANs, and Voice over IP (VoIP) is one of the most popular and promising among them. VoIP over WLAN (VoWLAN) is an innovative technology in the future of communication that has been quickly accepted by many health care administrators and care delivery staffs because it is cost effective, easily deployed and interference free.

In the purpose of reducing medical errors and provide better patient care services, communications in hospital become vital and time critical. VoWLAN fulfills the requirements of delivering critical information to relative professions. Moreover, it can also improve the efficiency, accuracy, and speed of care delivery and enhance management of the health system. The rest of this article is organized as follows. Section 2 introduces the actual state and architecture of the communication system in the hospital environment. The proposed hospital patient care call system and implementation are addressed in Section 3 and Section 4, respectively. Section 5 presents applications of medical professionals and section 6 concludes the work.

## 2. OVERVIEW

In Section II, we introduce the actual state and architecture of the communication system in the hospital environment.

### 2.1. Hospital Communication Approaches

In hospital, the actual state of the communication system can be divided into two categories according to the communication manners they are intended for. One is the direct communication, like oral communication between face-to-face conversations. The other is indirect way, including the written patient record, telecommunication and Internet communication. The face-to-face conversation has been demonstrated to influence patient outcomes, but it lacks flexibility and mobility. The written way is short of interaction among the care professionals. This disables health professions to handle the situations immediately and diminishes the health service. Telecommunication such as Public Switched Telephone Network (PSTN), pager, Personal Handy-phone System (PHS) is popular in most hospitals now. The PSTN and pager system bring the convenience but still cause the waiting times on hold and the traveling back and forth to a phone [1]. Moreover, PHS has the problems about the dead spot, junk mail and requiring recurring monthly. The primary disadvantage is that PHS system cannot rapidly deploy new service model to diverse medical situations. It is evident that the oral, written, and telecommunications cannot fulfill the requirements of delivering critical information to relative professions. The conversation via Internet, such as VoIP, VoWLAN can make up the deficiencies of communication ways as mentioned above. We have further introduction in the next section.

Nurses are the first-lined caregivers to offer immediate care to patients. If patients are in critical situations which requiring further treatments, nurses will then contact doctors on-duty first, then resident doctors, visiting doctors or relevant others in order. However, it is the generally routines that nurses cannot carry any communication devices that may result in interference to instruments and cause disturbances for patients. However, as their job duty, they have to work between wards and work-stations, and if anything happens on their way to wards or to units, they cannot offer assistance in time. Therefore, it is quite time consuming for the nurses to find out patients' needs and react properly according to patients' conditions.

In [5], authors proposed a medical vehicular to reducing medical errors. It provides efficiency and accuracy Unit Dose Delivery Service (UDDS) with RFID technology. This system provides mobility to care giving. However, it lacks effective communication modules and cannot meet the requirements of delivering critical information to relative professions.

## 2.2. Communication System Architecture

From above discussion, VoIP/VoWLAN services are sufficient enough to meet various settings. Wireless is a pervasive technology and has been widely used in most hospitals. Voice and data converged on the hospital network have become a prevalent communication trend to provide the optimal patient care service. **Figure 1** is the hospital communication architecture including PSTN system and novel IP telephony. An IP Private Branch Exchange (IP PBX) is used to combine Internet and PSTN for voice transmitting.

Voice Over IP (VoIP) is the delivery of voice traffic over the Internet. VoIP involves sending voice information in digital packets rather than by using the traditional PSTN [6]. The packets are transmitted using Real Time Protocol (RTP) over User Datagram Protocol (UDP) over Internet Protocol (IP). RTP is commonly used in Internet telephony applications. RTP combines the data transport with a Real Time Control Protocol (RTCP) to monitor data delivery.

Standards like Session Initiated Protocol (SIP) and H.323 are maturing to support convergence services in next-generation networks. H.323 was designed for multimedia communication over IP networks, including audio, video conferencing. SIP was design to be a part of the overall Internet Engineering Task Force (IETF) multimedia data and control architecture. However, many consider SIP a powerful alternative to H.323 because of its flexible deployment, easier maintainability, and simple format for commands, etc. As a result, a VoIP communication system based on SIP can develop diverse services correspond to different requirements.

There are two major roles in SIP, a client and a server. The Ward client sends a request to the server and then server passes the message onto Laboratory Medicine client. Via the connection setup, two clients can communicate with each other as shown in **Figure 2**.

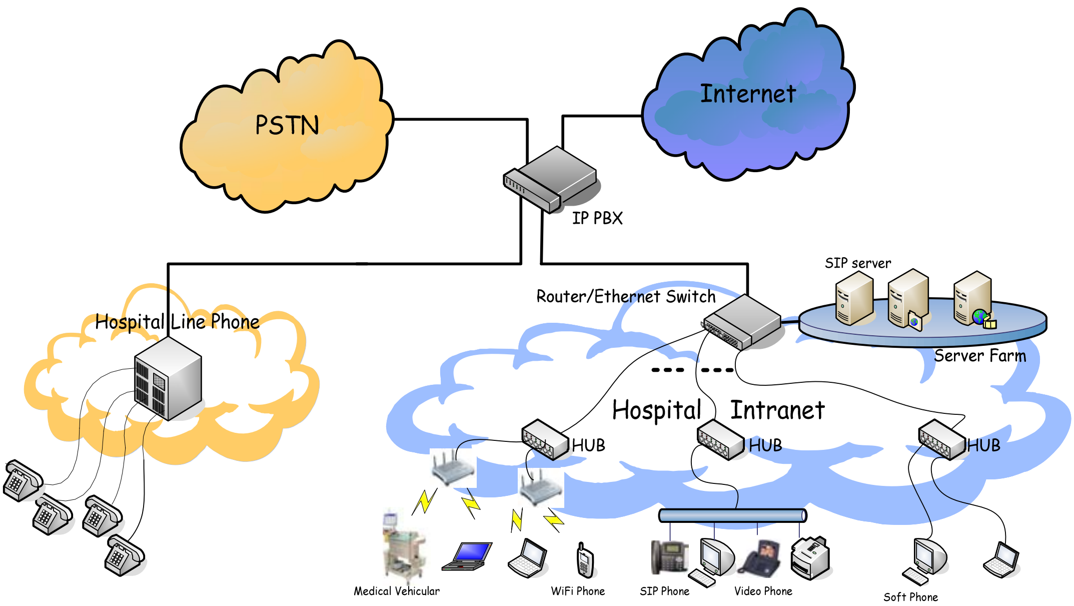


**Figure 1.** The hospital communication architecture includes PSTN and IP telephony system.

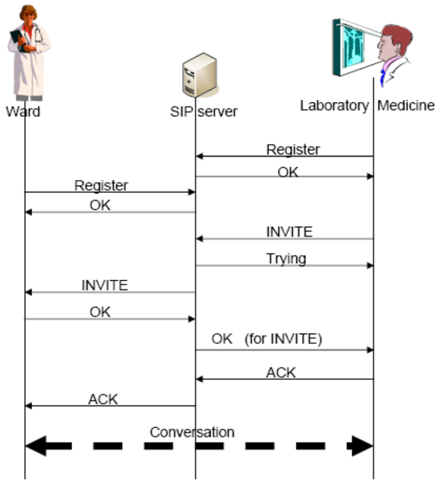


**Figure 2.** Signaling of a SIP call service.

## 3. HOSPITAL PATIENT CARE CALL SYSTEM (HPC2S)

According to the hospital communication requirements, we propose an intelligent and cost-effective communication system, called hospital patient care call system (HPC2S), to improve the quality of patient care. The main features of the proposed system are the one-number service and call forwarding.

One-number service means that a client can have multiple registrations with several terminals or devices. The client has a single number. When the number is called, the calls to the client may be sent to all registered destinations, such as office SIP phone, PC-based soft phone, and WiFi phone. Any of these destinations picks up the phone, server will automatically cancel others signaling.

Call forwarding service has two categories: conditional call forwarding and directive call forwarding. In conditional call forwarding mode, the server will transfer the INVITE signaling to another predefined destination if the signaling fails to contact anyone in a time interval. Apart from conditional one, directive call forwarding does not send INVITE signaling to the destination but rather forward to another predefined destination.

To illustrate the effective communication of HPC2S, a scenario and the signaling of SIP service are shown in **Figure 3** and **Figure 4**, respectively. Laboratory medicine needs to notify the doctor when the sample of the patient is abnormal. First they dial the mobile SIP hard phone (#333) of one doctor. According to one number service of the SIP server, proxy server will send signaling to both mobile SIP hard phone and the other SIP phone (#333) in the callee's office. Yet the callee had set the office phone the "Conditional Call Forwarding" to the phone of the ward (#222) earlier. SIP server will forward the INVITE signaling to the predefined destination automatically. It is one of the smart communication services of HPC2S, including one number service and call forwarding. Callers can have much more easier way to contact with callees, consequently save their time to check where callee would be and extension number.

## 4. IMPLEMENTATION

In this section, we introduce the design and implementation of HPC2S. HPC2S is developed based on SIP protocol. The detailed descriptions of the system architecture and implementation prototype are presented in the following subsection.

### 4.1. Session Initiation Protocol (SIP)

SIP is an application-layer protocol that can establish, modify and terminate interactive multimedia sessions, such as videoconference, Internet telephony. It is designed to address the functions of signaling and session management within IP networks. Signaling allows session information to be carried across internet. Session management provides the ability to control the features

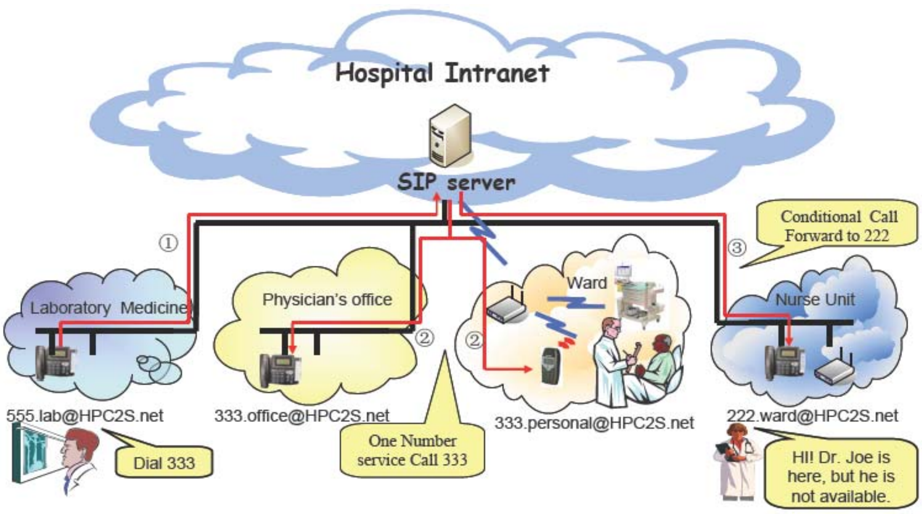


**Figure 3.** Calling flow chart of the One-number and Conditional Call Forwarding service.

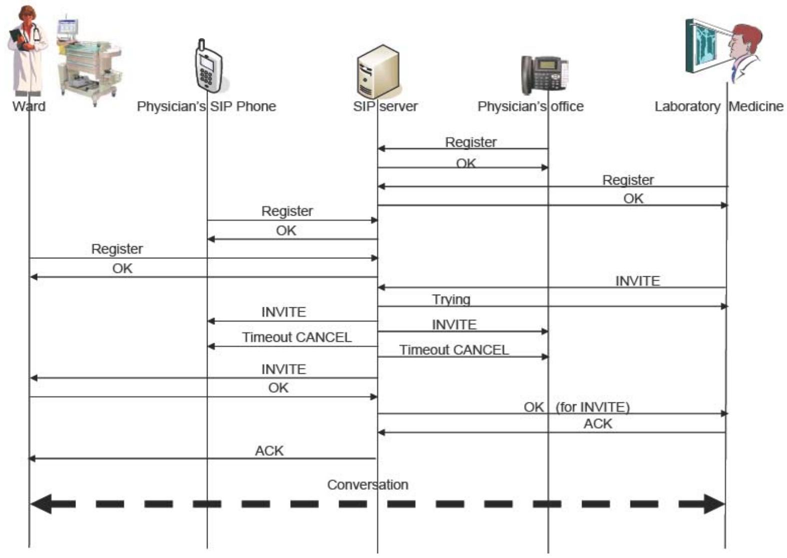


**Figure 4.** Signaling of a SIP One-number and Conditional Call Forwarding service.

of an end-to-end session. More accurately, SIP is a request-response protocol, dealing with requests fromclients and responses from servers. Participants are iden tified by SIP Uniform Resource Indicators (URIs). SIP server determines the end system to be used for the session, the communication media and media parameters, and the called party's desire to engage in the communication. Once these are assured, SIP server establishes session parameters of the communication, and handles session transfer and termination [7].

The SIP architecture can be grouped into two categories: clients and servers. SIP servers include proxy server, redirect server, registrar server. SIP clients include softphones, such as phone capabilities installed in personal computer, and IP phone. In addition, the SIP servers can interact with other application servers, such as location servers, to provide a wide range of value added services.

### 4.2. HPC2S Components

According to SIP protocol, the design of HPC2S is shown in **Figure 5**. One is the administration functions that include authentication, register, and configure for access control. The other is SIP proxy server. SIP users are identified by unique SIP addresses. Client needs to register at the SIP server first using their assigned SIP addresses. The authenticated clients also can update their information stored in SQL database with *UseDir* JAVA class. Other SQL database related JAVA classes, such as *IdImport*, *RegisteredList*, *SessionList*, and etc, provide useful functionalities for the web-based user interface. *IdImport* provides the registration for client and *RegisteredList* can show who have been registered. *SessionList* class is used to identify which sessions are proceeded. SIP server has diverse JAVA classes to communicate with SQL database. *ProxyServer*, SIP, and *SIPPacketMessage* resolve the SIP packets to create a session, such as INVITE request packet. In a session, *SessionMessageProcessor* class decomposes the session message to initialize *InviteRequest*, *RingingTimeout*, and *TalkingTimeout* according to received messages. Obviously, the HPC2S provides the administration setting and establishes and terminates VoIP calls.

### 4.3. SIP Call Establishment

When a user initiates a call, a SIP request is sent to a SIP proxy server. The request includes the address of the caller and the address of the intended callee. **Figure 6** shows the SIP point-to-point calls establishment process started with an INVITE message. The following are the procedure descriptions.

***Procedure Call-establishment***

***BEGIN***

***Step1*** ∶ *Caller sends INVITE request to SIP server, which extracts SIP Packet Message*

***Step2*** ∶ *Server sends INVITE request to callee.*

***Step3*** ∶ *Session message processor enables both ringing timer and talking timer, then ACK message and ringing signal are sent back to caller. Session message processor also needs to deal with all session packets.*

***Step4*** ∶ *Callee sends OK to session message processor.*

***Step5*** ∶ *Caller receives OK message.*

***Step6*** ∶ *ACK message is sent from caller.*

***Step7*** ∶ *ACK message received from callee.*

***Step8*** ∶ *Session is set up and they can access voice or media data transmission.*

***END***

Since a given client can have multiple registrations at a SIP proxy server. If a user has multiple active registrations, then calls to the user will be sent to all registered destinations based on the call-establishment procedure. This capability can enable the one number service. SIP proxy server can also change the Request-URI of an INVITE to the suitable contact addresses. If SIP proxy server does not receive the OK(INVITE) message, it will automatically transfer the call to the presetting address. It is evident that the call forwarding function is achieved.

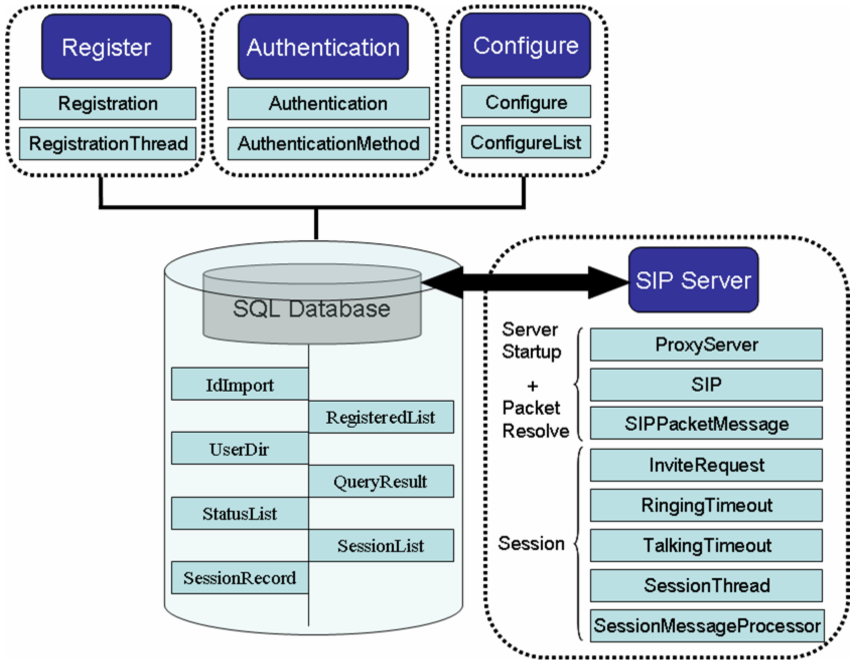


**Figure 5.** The function components of HPC2S.

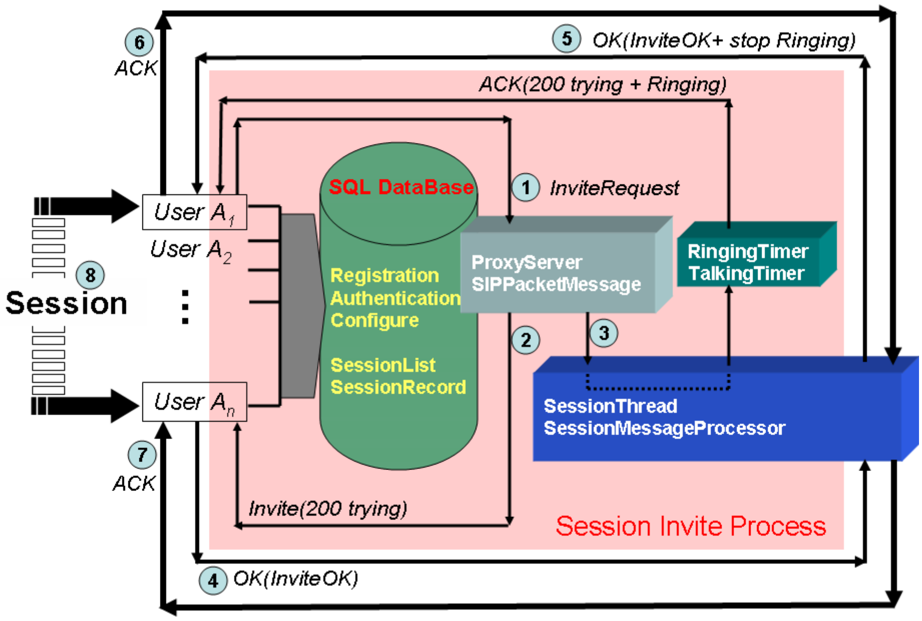


**Figure 6.** Process of SIP call establishment.

HPC2S both combines one number service and call forwarding to support convenient communications for the medical environments.

## 5. APPLICATIONS

The most powerful advantage of HPC2S is the flexibility to adapt to various hospital situations and even for diverse users. All the service developments not only meet the special needs of hospital users but also improve the quality of patient care. Here are some applications carried by HPC2S as shown below:

*A.* *Physicians*

- To increase efficiency in service delivering, physicians can immediately and directly converse with other health professions about lab results, further treatments, and patient updates.
- Physicians can locate related health professions as well as relevant nurses from the medical vehicular, and this can shorten the response time in finding the specific person.

*B.* *Patients*

- By using HPC2S, patients can reduce waiting time for caregivers and enhance the relationship between physicians and nurses.

- HPC2S bundled in medical vehicular can provide more delicate services with fewer pharmacological and other medical errors.

*C. Nurses*

- Nurses can offer high-quality patient care with medical vehicular and eliminate possible medical errors.
- HPC2S allows nurses to locate physicians conveniently, and can dramatically reduce required walking and holding time on the phone.

*D. Staffs*

- Staffs can have a conversation with the physicians and nurses at the same time when patients' examination results reveal abnormal.
- The hospital resources can be arranged more effectively via the HPC2S by staffs, such as utilizing operating rooms via better scheduling procedures.

## 6. CONCLUSIONS

In this article, we proposed a SIP-based hospital patient care call system. HPC2S provides a timely and effective communication among the care professionals to improve the quality of patient care. The medical vehicular with HPC2S have become more reliable and robust for delivering critical information. Nurses have direct conversations with other health professions by the intelligent medical vehicular. Nurses can also provide complete patient care without interferences. Since the network location awareness may have many potential applications. Therefore, how to enable network location awareness with HPC2S would be one of our future works.

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