Mobile IP Telephony: A solution to Rural Connectivity

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Abstract: With the development of wireless LAN and VoIP technology, economical build-up of infrastructure in the rural areas of developing countries is now emerging as a real possibility. The paper proposes the deployment of WiFi/WiMAX integrated with VoIP as a plausible solution to rural connectivity. Integration of the designed network with the existing cellular network has been suggested to extend the coverage range. Solar power dependency along with remote network management has been proposed to allow undisrupted connectivity. Simulations have been performed on the Network Simulator (NS2) to validate the efficiency of the proposed model. A comparison to other potential solutions to rural connectivity has been drawn to prove the benefits of the model.

1. Introduction

It is widely accepted that communication technology is one of the most important enablers, increasing access to information and thus the standard of living. Recent advances have greatly reduced the cost of telecommunications infrastructure and worldwide mobile phone penetration has increased greatly. However, most of the gains of the telecommunications revolution have been restricted to the industrialized countries. The central issue is shifting from disparity between developed and developing countries to between urban and rural areas of within developing countries.

Telecommunications carriers around the world have already introduced IP into their networks because it provides economic benefits over traditional telecommunications networks. Technologies that use the Internet and Internet protocol networks to deliver voice communications have the potential to reduce costs, support innovation, and improve access to communications services within developing countries and around the world. We have proposed the integration of WiMAX with VoIP to provide coverage to a large customer base.

We start by giving an overview of VoIP technology and WiMax. We then present a design of the model to be deployed. We continue by comparing of the proposedtechnology with the other viable solutions and how mobile IP telephony proves to be the most plausible option.

2. Overview of VoIP

The Public Switched Telephone Network (PSTN) is the collection of all the switching and networking equipment that belongs to the carriers that are involved in providing telephone service. In this context, the PSTN is primarily the wired telephone network and its access points to wireless networks, such as cellular. The overall technology requirements of an Internet Protocol (IP) telephony solution can be split into four categories: signaling, encoding, transport and gateway control. The purpose of the *signaling* protocol is to create and manage connections between endpoints, as well as the calls themselves. Next, when the conversation commences, the analog signal produced by the human voice needs to be *encoded* in a digital format suitable for transmission across an IP network. The IP network itself must then ensure that the real-time conversation is *transported* across the available media in a manner that produces acceptable voice quality. Finally, it may be necessary for the IP telephony system to be converted by a *gateway* to another format-either for interoperation with a different IP-based multimedia scheme or because the call is being placed onto the PSTN.

An IP network does not have the same limitations as the traditional telephone system as far as routing is concerned. A circuit switched network creates a dedicated connection for an entire call, and therefore guarantees a constant bandwidth and satisfactory quality for the entire speech transmission. A packet switched network like the Internet splits the voice data into packets that are routed independently along the most efficient path in the network on the way to their final destination.

SS7 Once a user dials a telephone number signaling is required to determine the status of the called party—*available* or *busy*—and to establish the call. Signaling System 7 (SS7) is the set of protocols (standards for signaling) used for call setup, teardown, and maintenance in the Public Switched Telephone Network (PSTN).SS7 is implemented as a packet-switched network and typically uses dedicated links, nodes and facilities. In general, it is a non-associated, common channel out-of-band signaling network allowing switches to communicate during a call. The integration of SS7 and IP will provide significant benefits**.** Figure1 depicts a type of VoIP network utilizing an SS7-to-IP gateway. SS7 provides the call control on either side of the traditional PSTN, while H.323/Session Initiation Protocol (SIP) provides call control in the IP network. The media gateway provides circuit-to-voice conversion.

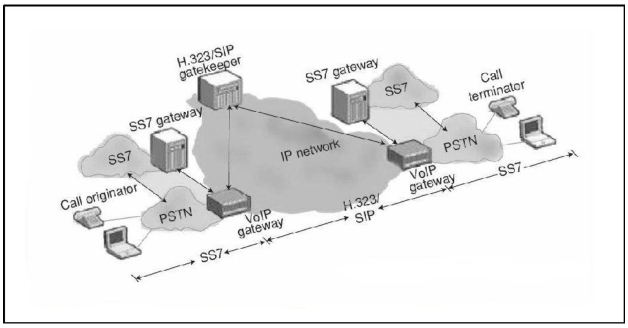


Figure1. SS7 based VoIP Network

H.323H.323, ratified by the International Telecommunication Union-Telecommunication (ITU-T) is a set of protocols for voice, video, and data conferencing over packet-based networks, such as the Internet. The H.323 protocol stack is designed to operate above the transport layer of the underlying network. The scope of H.323, however, is much broader and encompasses networking multipoint conferencing among terminals that support not only audio but also video and data communications. In a general H.323 implementation, three logical entities are required: gateways, gatekeepers and multipoint control units (MCUs).

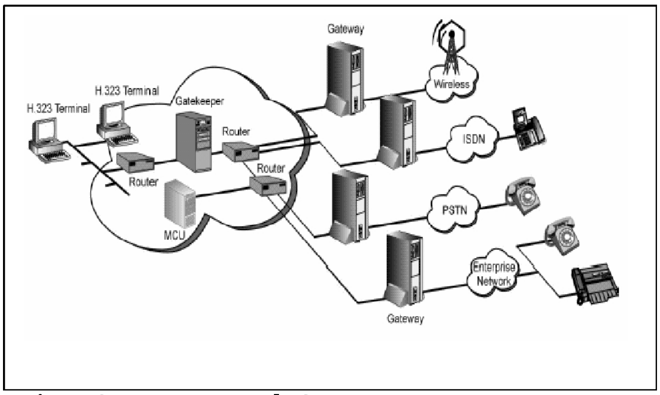


Figure2. VoIP Network Components

SIP Session Initiation Protocol, SIP, defined by the Internet Engineering Task Force (IETF), is a signaling protocol for telephone calls over IP. Unlike H.323, however, SIP was designed specifically for the Internet. It exploits the manageability of IP and makes developing a telephony application relatively simple. SIP is an application-layer control (signaling) protocol for creating, modifying and terminating sessions with one or more participants. The facilities of SIP enable personal mobility-the ability of end users to originate and receive calls and access subscribed telecommunication services on any terminal in any location. This mobility can be augmented via wireless VoIP.

Voice coders An efficient voice encoding and decoding mechanism is vital for using the packet-switched technology. The purpose of a voice coder (vocoder)-also referred to as a codec (coding/decoding)-is to use the analog signal (human speech) and transform and compress it into digital data. A number of factors must be taken into account including bandwidth usage, silence compression, intellectual property, look-ahead and frame size, resilience to loss, layered coding, and fixed-point vs. floating point digital signal processor (DSPs). The bit-rate of available narrowband vocoders ranges from 1.2 to 64 kbps.

Transport Once signaling and encoding occur, Real-time Transport Protocol (RTP) and R e a l - T i m e Control Protocol (RTCP) are utilized to move the voice packets. Media streams are p a c k e t i z e d according to a predefined format. RTP provides delivery monitoring of its payload types through sequencing and time stamping. RTCP offers insight on the performance and behavior of the media stream, such as voice stream jitter. RTP and RTCP are intended to be independent of the signaling protocol, encoding schemes, and network layers implemented.

Gateway In VoIP systems gateways function as the interconnection between the Internet and the PSTN or the H.323 and the non-H.323 network. On one side it connects to traditional voice signals, and on the other side it connects to packet-based devices. The gateway translates signaling messages between the two sides with a CODEC function and can also function as a compressor and decompressor.

3. WiMAX

WiMAX (Worldwide Interoperability for Microwave Access) is an upcoming new technology that delivers high-speed, wireless broadband at a much lower cost than cellular and over a much greater range than WiFi. WiMAX will not only deliver significant improvements in speed, throughput and capacity but will also enable portable and mobile services to laptops and handheld devices over a wider area of coverage making them more “mobile”.

The greater operational range of WiMAX provides our model significant advantages over the traditional wireless networks and the Wifi networks since operation on the widespread rural geography is primarily governed by the cost and coverage factors. The greater speed capability of the WiMAX network enhances the service quality of the VoIP phones which is one of the key issues wile evaluating VoIP system performance . Along with the support for high speed broadband , WiMAX offers the best possible combination of services that are needed for a rural setup ranging over large physical areas and hence is our choice in the proposed system .

4. Proposed Model for the Rural Connectivity

The figure 3 reflects a design of the model to be implemented in the rural areas to provide connectivity within the community as well as outside the community. The core technologies for the proposed Community Telecommunications network are primarily; 1) an Internet Protocol (IP) network in lieu of a circuit switched network 2) Voice services that are provided through Voice over IP (VoIP) in lieu of custom hardware-based switching 3) Wireless distribution, be it WiFi and/or WiMAX in lieu of terrestrial land lines. Together these newer technologies provide a unique opportunity that is just now capable of being realized, for expanding communications into the most remote areas of any country in the world. Further, these off-the-shelf technologies allow this to be done at a cost that is literally pennies-on-the-dollar for what has been possible in the past.

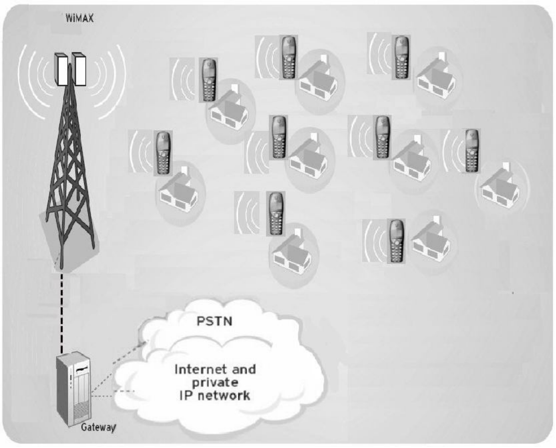


Figure 3. Model of the Rural Connectivity

Local Wireless Network/WiMAX As conceived and tested, the local community wireless network is comprised of a switch (or access to a central switch) a single router, a series of wireless antennas and repeaters, and VoIP phones, and in most cases, WiFi phones. The type, number, and placement of the antennas is a factor of the local environment, and may include WiMAX for distribution and WiFi for access, or may include a pure WiFi-only network, possibly a mesh.

VoIP Switching/Gateway This component is the actual VoIP switching that allows for calls to be routed between the user communities, whether they are within the local community or with those residing on other networks including other similar IP-based community networks, mobile users, or PSTN users. As proposed, the configuration is such that a single VoIP switch can provide support to hundreds, if not thousands of local community phone systems/customers. In fact, as conceived, a single soft switch can provide the needed switching for hundreds, even thousands of community phone systems located in dozens of different countries. It is equally feasible for each country to set up their own central VoIP switch to handle the local community networks within each specific country. Siemens offers its communications systems solutions, HiPath which is currently being used in smaller networks to facilitate VoIP and WiFi.

VoWiFi Phones With the growing populating of VoIP switching and WiFi networks, the most recent technological development has been SIP-based VoIP/WiFi phones. These look very similar in size to a mobile phone offered through any number of commercial wireless carriers. In fact, one of the current dynamics within the cellular marketplace is the dual-mode handsets that provide digital cellular service as well as VoWiFi capabilities, including switching between the two networks while retaining a call. The advantage is that they can be powered by solar where needed, and can be operated directly off of the community WiFi network.

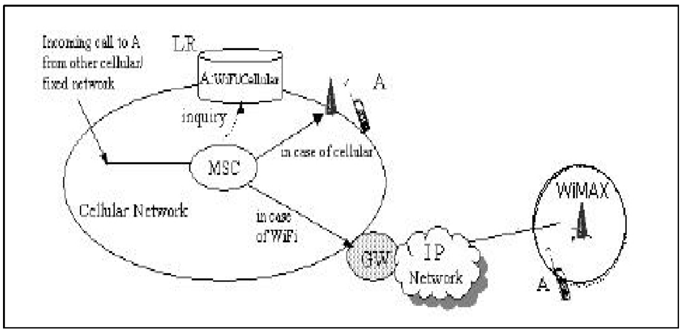


Figure 4. Integration of Cellular and WiMAX Network

Integration of cellular and Wi-Fi networks

Figure 4 shows the method where wireless IP phone is assigned a cellular number, and when it is outside cellular coverage area, it is connected through fixed IP network. This enables maintenance of the connection while moving out of the coverage of the cellular region into the rural region with WiFi coverage.

In cellular network, HLR (Home Location Register) keeps the area where each subscriber is located (the location area). When the subscriber moves and the area changes, the HLR is also updated to a new area code. As one of these location areas, "home Wi-Fi area" is added With this system, when in the 'hot spots', connection via VoIP is possible. Outside, it is connected via existing cellular network. When the terminal detects Wi-Fi radio signal over certain threshold, it notifies the cellular network through IP network that it is located in its own "home Wi-Fi area." GW in Fig.3 converts the telephone number to IP address, and connects to a designated access point.

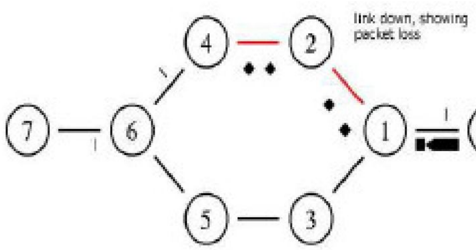
Securing power source Commercial power supply is pre-requisite for location of Gateway (IP-PBX), but location of access points in Wi-Fi needs to anticipate the possibility of no power, these do not require high power like traditional telephone switching systems did. In places without power, solar power with simple solar panel or small, light-weight Manganese Lithium-ion type batteries can be used. For the

VoWiFi phones, portable solar power charger is commercially available.

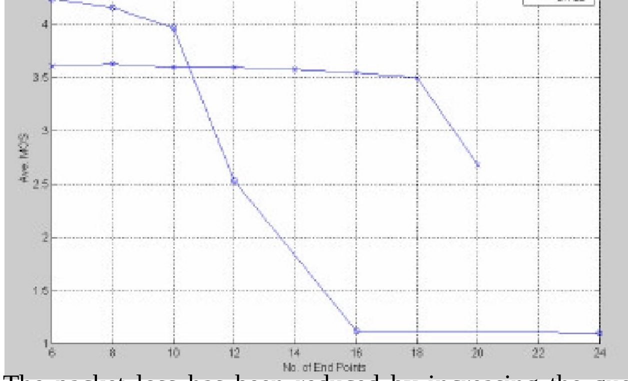
Remote maintenance capability Since it is rural area, remote monitoring and control function is indispensable. The local systems from all the communities can be connected to central control centre via Internet VPN, in order to collect data on operation to be able to cope with problems if they arise. When large-scale WiFi network or WiMAX is implemented in rural areas and they are connected to public cellular and fixed networks, an integrated network management system will be needed. For this type of network management, SNMP (Simple Network Management Protocol) may be applied to wireless LAN devices such as access points, thereby enabling build-up of a management system easily in a short period of time.

5. Simulation Results

To test the efficiency of the proposed communication model we simulated the same on the Network Simulator (NS2) and arrived at the following results.



The capacity of the two different codes G.711 and G.723 are compared and it is concluded that 802.16 (WiMax) protocol can support can support fourteen to eighteen simultaneous VoIP sessions using the G.723.1 codec.



The packet loss has been reduced by increasing the que length which though increases the packet arrival rate by some amount. Future work is to optimize the packet arrival rate and also reduce the packet loss.

6. Potential limitations of IP Telephony

Despite its several benefits there are a few potentials limitations to the delivery of voice over packets. Some of them are discussed here.

Delay also called the latency is the time taken for a packet to arrive at its destination from the sending endpoint. The IP network delay is the sum of the packet capture delay, the switching/routing delay, and the queuing time.

Jitter Packet delay variation is called jitter and is defined as the variation of packet inter-arrival time at the receiving side of the network. In VoIP systems, jitter is also produced when packets are sent over different paths through the network. Some packets may therefore arrive in a different order compared to how they were sent. Jitter is a more or less unavoidable factor in networks dealing with multimedia services.

Packet loss corresponds to the percentage of the total amount speech frames that do not reach their point of destination. It occurs frequently in IP connections and can have many different origins. Packet loss may be caused by physical media errors as well as an overloaded router that intentionally discard packets to relieve congestion in the connection. A packet can also arrive too late to be involved in the reconstruction of the voice signal and therefore get dropped. Packet loss can in turn create impairments like dropouts and time clipping.

Echo : Echo is caused by the leakage of the talkers voice signal between the transmit path and the receiving path, often caused by a mismatch between the analogue telephone device and the transmission media. Another reason is acoustic coupling problems between a telephones microphone and its loudspeaker.

7. Comparison to CDMA450

CDMA 450MHz is often proposed as a rural WLL solution, based on the idea that propagation at 450MHz is superior; therefore coverage using this technology can be achieved very economically in rural areas. However, it is critical, for operators evaluating this technology, to understand that range in CDMA systems is limited by noise and mutual interference, not by propagation and path loss. CDMA 450MHz can achieve 50km range, but in single-user conditions, or under very light system loading. When the base stations are sufficiently loaded to amortize their cost, the achievable range will be much less. Typically, one base station must be installed in each served community, even if the network operates at 450MHz. Moreover, being a mobile network, CDMA-450 requires a complex core network mandated by mobility standards.

Operators of fixed rural CDMA networks will find that they have to:

• Over-invest in the CDMA core network
• Integrate mobile network nodes into their fixed line networks
• Build extensive, high-performance backhaul networks
• Deploy an excessive number of base stations
• Use dedicated, high-power phones and directional antennas for remote subscribers

The result is that a rural CDMA-450 network can be much more expensive than operators would expect based on the nominal traffic capacity required by the network. Mobile IP telephony systems like the one which we proposed are much more economical to deploy in rural areas, and are particularly more cost-efficient to expand, because of their stable coverage under increasing traffic conditions.

8. Conclusion

We have made a proposal for a rural telecommunication system targeted at expansion of telecom infrastructure in rural areas of developing countries, utilizing VoIP, WiFi/WiMAX and cellular/Wi-Fi dual terminals. This approach is substantially less costly than reliance on traditional circuit -switched solutions and even cellular networks.

This proposal which integrates cellular and WiFi networks on VoIP attempts to have the large-scale, nationwide cellular network and local, small-scale wireless LAN collaborate and fuse with each other, and it provides superior convenience. It is possible to implement in a short period of time, and it is provides the capability to cope with localities with no power supply, and remote maintenance. Furthermore, through using the Internet over this infrastructure, people will be able to develop various applications for tourism, environmental protection, education and government services that are needed in each local community.

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