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Rural Telemedicine Networks Using Store-and-Forward Voice-over-IP[1]

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Abstract. Store and forward Voice-over-IP is a suggested solution for supporting Telemedicine at rural health clinics in developing countries. Solutions described to date are designed to support communication by establishing point-to-point connectivity between two sites. In this paper we present an approach for creating scalable Telemedicine networks based on Delay Tolerant Networking. This holds potential for allowing Telemedicine networks to be created that can enable sharing of Teleconsultation and other medical information among a large number of locations in areas that cannot be served by existing solutions.

Keywords. telemedicine, Voice-over-IP, delay tolerant networking

1. Introduction

Telemedicine has been noted as a potential way to improve healthcare in rural areas of the developing world [1]. There are several barriers to introducing ICT solutions in this domain including cost, poor infrastructure and a lack of computer skills among staff [1]. Rural healthcare providers in many areas of the developing world also lack connectivity to the Internet and the Public Switched Telephone Network (PSTN).

Design studies and pilot projects at clinics in South Africa suggest that one approach for dealing with these challenges is to utilize store-and-forward Voice-over-IP (VoIP) [2, 3]. VoIP is accessible from devices similar to traditional telephones, and thus can be used by those that lack computer skills. VoIP services can also be developed relatively cheaply using open source software, and can be deployed without fixed infrastructure (such as by using point-to-point WiFi links) or support from a telecom provider [4]. The rationale for store-and-forward services (instead of relying on standard voice calls) is that they operate effectively during periods of disconnectivity, and also allow staff to manage interruptions more easily [2, 3].

This paper presents issues related to the development of rural Telemedicine networks based on store-and-forward VoIP. Issues related to the development of the relevant end-points and network infrastructure are presented. As a similar approach

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may also be used for general data transmission, the paper may also be of interest for those investigating other store-and-forward Telemedicine services in rural areas.

2. Communication Technologies for Low Connectivity Areas

Point-to-point WiFi links using directional antennas are one way of providing VoIP services in rural areas [4]. This approach has been applied to "MuTi", a Telemedicine system for rural South Africa [2, 3]. Although there are many settings where point-to-point WiFi will function well, there also are situations where such connectivity is not ideal; it is not always possible to maintain line of sight between two communication points (i.e., due to distance or interference) and directional antennas are somewhat expensive. Setting up a large VoIP network based on this solution may thus prove challenging due to the additional coordination, cost and maintenance work involved with implementing each location.

The approach we advocate in this paper is to build store-and-forward VoIP networks based on Delay Tolerant Networking (DTN) [5]. DTN operates by leveraging mobility and local communication between participants in the network. Each member of the network communicates with other members when possible, for example when they are close enough for local wireless communication (using WiFi, Bluetooth, etc.), or when a long range link becomes available. Members store messages from each other and forward them later on when they establish connectivity with other members [6]. Each act of forwarding can help messages get closer to reaching their final destination. This allows communication of non time-critical information between participants in the network, even if there is never a completely connected path between them and in practice has shown to be effective at supporting fairly large communication networks.

Over the past few years, a large amount of progress has been made on solving many of the basic technical problems related to DTN including routing [7, 8] and privacy and security [9]. Services using DTN have also been implemented in various contexts. One of the first services to utilize DTN technology established Internet services for reindeer herders in Northern Sweden [10]. Connectivity was provided primarily by using wireless devices carried by hikers, helicopters, and other vehicles. These devices support DTN routing, and exploit the semi-random encounters of the mobile participants, and a history of previous encounters, to pass messages through the network in an efficient manner. Messages are forwarded during an encounter between two participants when there is enough probability that such forwarding will help the message eventually reach its destination [8]. This has allowed services such as email, cached web access, and basic file and data transfer to be supported, even though Internet, PSTN and other connectivity options are not available.

DTN has also been investigated for Teleconsultation services in Africa [11], and has been suggested as a component of store-and-forward voice messaging services for the "next billion users" of mobile phone technology [12]. In this respect DTN based voice messaging services could be used for Telemedicine services at rural health clinics.

3. Implementing a VoIP/DTN Network

There are many approaches that can be employed when setting up routing in a DTN network. One approach is to use dial-up links that connect once or twice a day. In such networks, optimal routing is fairly easy to achieve using linear programming approaches [13] or some scheduling mechanism. In most cases link availability may have a random element in it, and the most efficient path from source to destination also may vary between different times. This may be the case for example if the DTN relies on devices carried by health care workers with variable travel schedules or long range links that are only available when the weather is good or power is available.

Initial work on data delivery in DTN networks focused on *epidemic* routing of data where the spread of data is modeled as the spread of a disease in a human population [14], and the data is eventually spread to the entire network. Later work, has focused on trying to estimate the mobility characteristics of the network participants, and on using that information to improve message routing and reduce the cost of the system.

3.1. Developing VoIP Services over DTN Tunnels

Our proposed solution involves providing VoIP services over DTN tunnels based on a store-and-forward approach for delivering voicemail messages. The components of the system proposed are depicted in the architecture shown below (Figure 1).

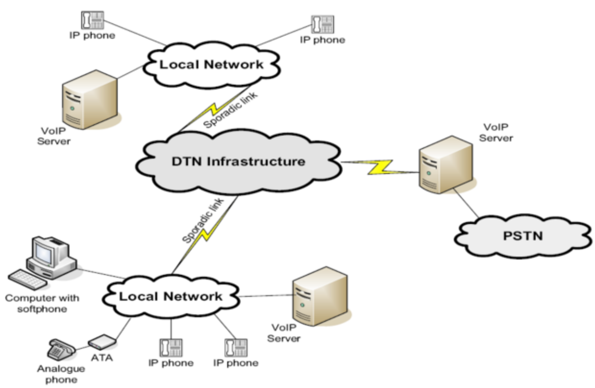

Figure 1. The VoIP/DTN architecture

The scenario involves providing the users with a VoIP phone. This could either be software running on a computer (softphone), an IP phone, or a regular phone attached to an Analog Terminal Adaptor (ATA) that allows it to connect to the VoIP server (offering IP phone functionality). The phones connect to a local VoIP server which operates using the Session Initiation Protocol (SIP) protocol. A user dials a destination number as he/she would for a normal call. Since connectivity is not available, the user is given the option of leaving a voice message.

This solution allows hardware costs to be kept to a minimum. DTN routing can be supported by inexpensive hardware such as simple Bluetooth devices. VoIP servers can run on fairly inexpensive hardware, such as used Pentium IV PCs. Additional hardware (such as a keyboard, monitor, mouse and headset) can be used to provide a softphone client extension. IP phones can be used for multiple client extensions at each site (eliminating the need for multiple computers) where a VoIP server is installed. Cheaper solutions should also be possible in the future as inexpensive mobile computers become powerful enough to support a VoIP server.

Traditional Internet protocols (including those used by VoIP servers) do not work in DTN environments since they tend to be very "chatty" and exchange many messages between endpoints within a single session. Therefore, the Bundle Protocol was created for DTNs, where the basic message unit is a *bundle*. The idea is to package all messages from an entire session into a single bundle; this can be sent over the DTN using application specific bundling gateways on both sides of the DTN.

Transmitting VoIP data onto the DTN network thus requires the creation of a VoIP/DTN tunnel that can *bundle* packets, transmit them to the DTN network, and *unbundle* these packets at their destination [15]. For VoIP the receiver's destination (i.e., the DTN endpoint) is identified based on the SIP URI of the called party. The Distributed Universal Number Discovery protocol (DUNDI) [16] can be used to establish peering information between the distributed VoIP servers. It is also possible to support additional data such as video and images by providing the appropriate devices, and by bundling the data and transferring it onto the DTN network.

3.2. Delivering DTN-Based Voicemail

Many of the basic components that are necessary for a VoIP/DTN system are available via open source libraries. This makes it possible to build VoIP/DTN systems without investing in proprietary software. One of the most popular VoIP servers is Asterisk [17]. Below we discuss how to bundle and debundle Asterisk voicemails onto a DTN network, but solutions based on other SIP-based servers are equally applicable.

When voice messages are created on an Asterisk server there is an option of having an external application launched by using the *externnotify* command. This allows for the tunneling mechanism to be activated and the DTN bundle to be created. A fairly straight forward way to support bundling is to use the file copying tool for file transfers over the bundle protocol. This will allow the Asterisk server to save the message envelope (containing information such as timestamp, sender, recipient and message length) and the audio file as a single file. The resulting file is then made into a bundle by the DTN software and is addressed to the appropriate endpoint.

The bundled voicemail is stored in a location where it will be accessed when contact is made with a DTN router. The bundle is forwarded to its corresponding endpoint (i.e., the destination server) using a DTN network mechanism as described above. At this stage, the bundle is extracted and the message envelope file and the audio file containing the message are placed in the recipient's incoming voicemail directory. The system generates the corresponding voicemail notification to the recipient's phone lighting the Message Waiting Indicator button (if available), Depending on the setup of the server, services such as emailing the voicemail to the recipient may also be provided.

Calls to and from the PSTN may also be supported by running a VoIP server with DTN and PSTN connectivity. Incoming messages would be handled similarly to regular voice messaging. Outgoing messages would be delivered by the server placing a call and playing back the recorded voice message after someone answers the phone. If necessary support for time-critical information may also be possible within the network by, for example, supplementing DTN communication with a satellite link. This however would create significant cost for time-critical messages.

4. Conclusions

VoIP/DTN networks that support store-and-forward voice messaging seem to be a promising approach for constructing rural Telemedicine networks in the developing world. The solution is not dependant on telecom providers and can be set up to support communication between many locations. Messages can be delivered during long periods of disconnectivity, and communication does not require a direct line of sight. Systems can be developed based primarily on open source software, and can be run on fairly inexpensive hardware, in order to keep costs to a minimum. We have described how to build a system based on existing open source components for VoIP and DTN in addition to a tunneling mechanism for transferring Asterisk voicemails to and from a DTN network. We expect to conduct an empirical evaluation of a system of this type in the near future. In addition to clinical evaluations there are a large number of socio-technical issues that need to be addressed before DTN-based solutions can give maximum value to health care systems without requiring support and maintenance from outside experts. The potential benefits of VoIP/DTN systems and open questions regarding their implementation suggest that VoIP/DTN Telemedicine systems should be an active area of investigation in the future. In addition, DTN-based Telemedicine solutions that support other data, such as data backups of Electronic Medical Records, may also provide benefits to rural healthcare systems and should be worth investigating.

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