A Delay Tolerant Platform for Voice Message Delivery

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Abstract - **A voice conversation is predominantly known as a synchronous, delay intolerant form of communication. To complement voice conversations, voice messaging is a form of asynchronous communication used when the called party is unavailable; a delay in message delivery is naturally expected and tolerated. To this extent, a Delay Tolerant Network can be utilized as the underlying data transport mechanism for the delivery of voice messages and in this paper we describe an architecture that combines this technology with the functionality of Voice-over-IP based systems. Our target application concerns the delivery of voice messages that originate from IP-based systems and are destined either for users of other IP-based systems or the Public Switched Telephone Network.**

Keywords: Delay Tolerant Network, voice message, VoIP, SIP

I. INTRODUCTION

Even in today's modern era where text messaging is the predominant way of exchanging small 'pieces' of information, voice messaging is still popular owing to its more personal touch and relative ease of use. Due to the fact that an immediate response (if any) is usually not expected, voice messaging can be considered as an asynchronous form of communication. In a fashion similar to email, the sender of the message does not know when the intended recipient will actually receive the message; the message can be stored on an answering machine for minutes, hours or even days.

In addition to the analogue and mobile telephony world, voice messaging is also used in VoIP-based (i.e. packet switched) systems, where the system can also be configured to send the message as an email attachment to the user associated with the number called; this is primarily done in order to expedite delivery of the message (e.g. the user is not near his/her phone but has email access) but the email may also not be readily accessible.

Based on the above and due to its delay tolerance, voice messaging is perfectly suitable for networks with intermittent connectivity. An intermediary gateway is often used to interconnect VoIP systems to the Public Switched Telephone Network (PSTN). In areas where connectivity occurs opportunistically or resources (e.g. telephone lines) are limited, it may not always be possible to place outbound calls and talk to the other party; in such a case, we propose that the caller leaves a message instead. Note that our target application is slightly different than traditional voice messaging where messages are stored immediately at the recipient's voice mail box and access time depends solely on the recipient; we have another inherent delay due to the frequent delivery network disconnectivity but at least, a messaging service is provided where it would not normally be possible.

In the architecture proposed in this paper it is assumed that when a call is placed, an end-to-end path from caller to recipient does not exist and a voice message will be left. Using delay tolerant network mechanisms, the voice message will eventually reach a system that can deliver it to its recipient. In our work, in addition to an underlying DTN architecture, we utilize an IP-PBX for call management, voicemail creation and PSTN gateway access.

In the following sections of this paper, we first give an overview of related technologies and then present our system architecture, its components and their functionality. Finally, we discuss our conclusions and plans for future work.

II. BACKGROUND

In this section we briefly present some background information on the technology and the tools our work is based upon.

A. Delay Tolerant Networking (DTN)

In order to facilitate communication between nodes residing in networks where some fundamental assumptions required for TCP/IP to work properly are not valid, researchers proposed [1] Delay Tolerant Networking (DTN). A DTN can be implemented in the form of a network overlay [2] that delivers packets using asynchronous message forwarding. It is designed to operate in environments with limited expectations of end-to-end connectivity and node resources; to this extent, it was initially proposed to facilitate interplanetary communication but as we will see it has also been useful in other applications.

Briefly put, DTNs can utilise a store-and-forward approach for data delivery. Mobile nodes within a DTN network act as mules; they store received messages and hold them until they "carry" them over to the next hop on their way to their destination. Before leaving its source, data is assembled into 'bundles'; the bundle protocol [3] is an application layer protocol that manages bundle propagation with the DTN area. To interconnect a DTN with the internet infrastructure and enable the delivery and reception of bundles, a "Convergence Layer Adapter" (CLA) is used. The CLA essentially acts as an intermediary between a DTN application and the IP protocol. Different CLA implementations offer varying services to the applications or try to optimize a particular form of communication; the most common examples are the TCP Convergence Layer Protocol [4], Saratoga [5] that mimics UDP and the DTN session layer [6] which is optimized for receiver-driven applications and multicast communication.

One of the biggest challenges in DTN applications is bundle routing; since connectivity is unpredictable, a next-hop is not usually readily available. Initially, the dominant routing technique applied was based on epidemic routing [7] which essentially follows a message flooding paradigm. However, different approaches have been proposed, each one optimizing a different aspect of routing performance. These approaches can be split into two categories: the first category contains approaches that improve the probability of delivering a message, and the second contains approaches that improve the store and forward process. In the first category, suggested algorithms calculate metrics (such as delivery probability) based upon which the best possible link [8] or path [9] is selected. The second category is more concerned with improving the aspect of bundle propagation; this can possibly be achieved by smarter caching [10] or by identifying and using the path with the lowest expected transit delay [11].

Finally, one can find in the literature different results derived from the empirical evaluation of DTN routing approaches. Simulations conducted by Lindgren and Phanse [12] reveal the clear benefit of using delivery predictability calculations instead of naïve epidemic routing. The evaluation of different DTN routing approaches in [13] showed that the performance of the routing algorithms in DTN is related to the amount of information the routing protocol has about the environment it operates in and the application it generated the data to be transferred; as the information provided to the algorithm increases, its performance improves. These results can lead to the conclusion that routing algorithms in DTN should be application and environment specific. One example that follows this paradigm is MaxProp [14] which is fine tuned for vehicular-based DTNs.

As mentioned above, DTN was initially proposed to support interplanetary communication. Wood et. al. proposed Saratoga [5], a DTN convergence layer that intends to provide a single flow between two peers with optimized link utilization. Saratoga was later used [15] for downloading earth images from satellites orbiting around the earth.

Realising the potential of this emerging technology, there were proposals [16, 17] for the use of DTN for connecting regions where high speed wired or wireless networks are not easily deployable or economically viable. Both papers present an architecture that supports connectivity of such regions to the Internet. An empirical study was also presented in [17] that showed the effectiveness of the proposed approach and its positive effects on people in poor and rural areas. Seth et al. [18] utilized DTN for providing low cost and reliable connectivity to designated areas (rural kiosks) in developing countries, offering a variety of "online" services to local people; another use proposed was for the provision of telemedicine services [19].

DTN was also utilized in the CarTel project [20] where a distributed, mobile sensor network and telematics system was developed for storing and forwarding human mobility traces collected in real-world environment settings [21]. In a more simplistic fashion, DTNs were also used for the delivery of emails: first within a DTN, then from the Internet to a DTN, and then from a DTN to the Internet [22]. Finally, in [23], the proposed system provides walkie-talkie functionality (using recorded voice messages) for communication between users residing in a DTN region. This work is based on a similar concept as ours but the system was designed for mobile users and messaging functionality is provided by running a special client on a mobile phone.

B. *Voice-over-IP and Private Branch eXchange systems*

At the application level, Voice-over-IP is a technology based on packet-switching that is nowadays standardized and easily accessible. The Session Initiation Protocol (SIP) [24] is the established standard for setting up VoIP calls and the information it conveys during the call setup process is vital for our application.

Many end user devices (IP phones) as well as software tools (soft phones) implement SIP; along with a range of SIP compliant IP-PBXs they constitute a fully fledged telephony solution. The IP-PBX systems support a vast number of features that range from simple voice messaging to Interactive Voice Response (IVR), Conference Calling, and Automatic Call Distribution. A powerful feature (found in open source PBX platforms such as Asterisk [25]) is also the ability to extend the functionality and call handling mechanisms.

In our work, we are using the Asterisk PBX which is an open source solution that in addition to the advanced functionality it offers can interface with PSTN gateways and more importantly allows user-defined programs to be launched to control telephony operations.

C. *The Distributed Universal Number Discovery protocol*

The Distributed Universal Number Discovery (DUNDi) protocol [26], provides directory services necessary for discovering routes to telephone numbers over the Internet. These telephone numbers can be in the form of local extension (e.g. an enterprise using four digit dialing plan) allowing calls inside an enterprise or in the form of E.164 numbers, permitting calls to real phone numbers available on the Internet. In addition to the above, DUNDi can be used for discovering a variety of information, including SIP directories.

We use DUNDi, which unlike other centralized technologies offering similar services (e.g. ENUM), is based on the peer-to-peer paradigm, requiring no centralized authority and having no single point of failure. Internet to Telephony Services Gateways can utilize DUNDi by sharing their dial plans and forming a trusted group within which DUNDi queries are sent and readily resolved. This feature is considered fundamental in our system, considering that we want to support dynamic discovery of user destinations.

III. SYSTEM ARCHITECTURE

Combining the technologies and tools described in the previous section, we defined the architecture of a system that utilizes voicemails to facilitate communication of users residing in isolated and disconnected geographical regions with the "outside" world. More precisely, in regions where communication through live phone calls is not feasible due to different network or capacity limitations, the proposed system records users' calls and attempts to deliver them to their intended recipients in the form of voicemails.

In order to achieve this and allow recorded messages to exit the disconnected geographical regions, the system relies on a Delay Tolerant Network infrastructure for the storage and forwarding of recorded voicemails. Finally, it is important to note that the proposed system is transparent to the end users; callers place regular phone calls and leave regular voicemail messages and callees receive regular voicemail messages.

The architecture of the proposed system is depicted in Figure 1. Below, we present the system's entities along with a detailed description of their functionality.

A. DTN Voicemail Station

The DTN Voicemail station (DTNVM) is in general terms the entity responsible for processing calls originating from users inside a DTN area (i.e., disconnected region), recording the voice messages and initiating their delivery procedure. It consists of two modules: the IP-PBX and the Voicemail Sender.

The role of the IP-PBX is to process a user's call request; in the case that connectivity is not available (which should be the norm in a DTN region) instead of dropping the call, it should prompt the user to leave a voice message. After successfully recording the voice message, the PBX creates a special file that contains the actual message along with destination information which is automatically extracted from the call setup information.

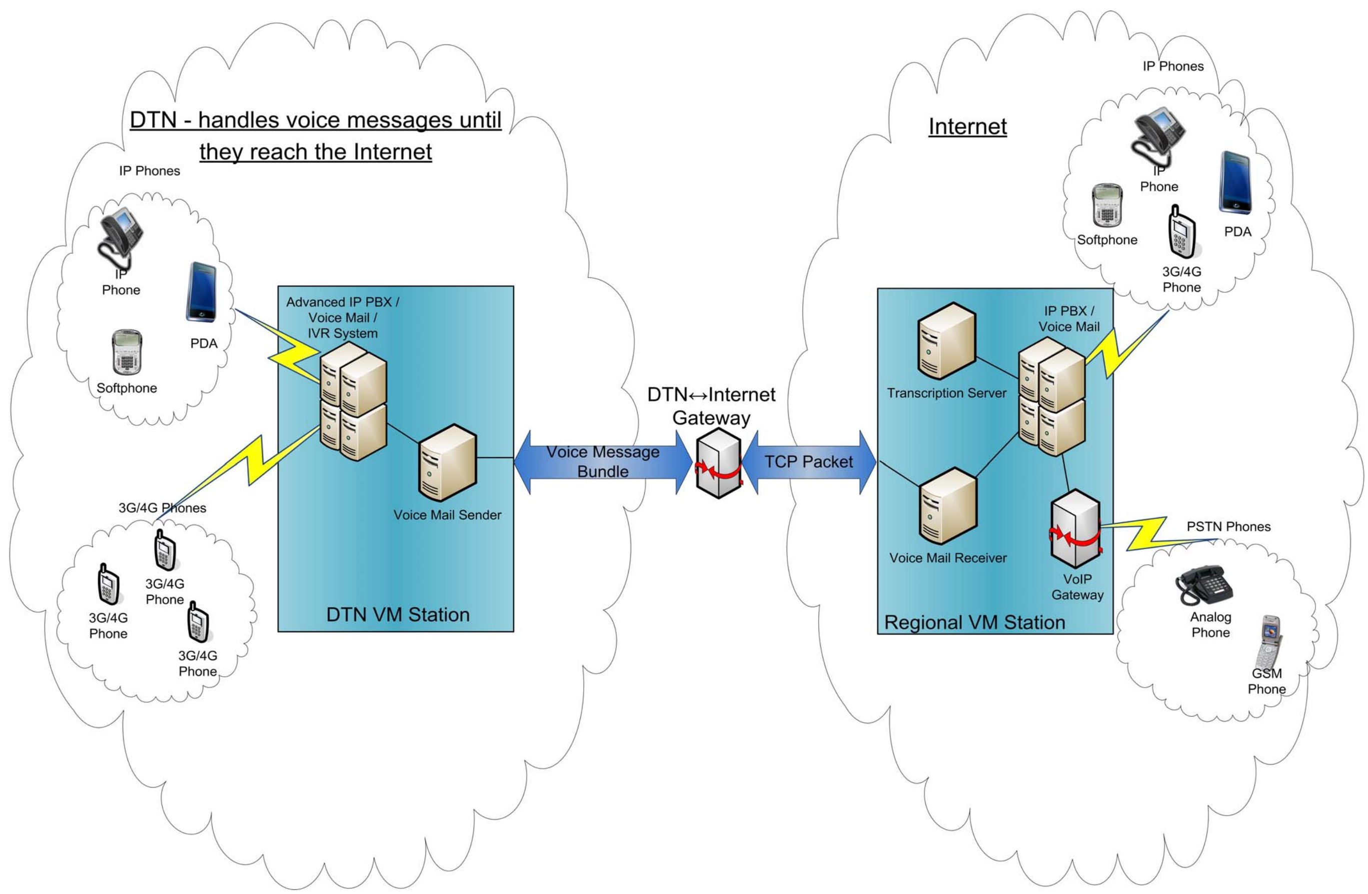

Figure 1. System Architecture

Once all the required information is in place, the DTN VM Station starts the process of forwarding the recorded voicemail to its intended destination over the DTN. The Voicemail Sender is the component responsible for retrieving the voice messages and using their destination information to initiate their delivery. The message recipient can be a SIP user (in the form of a SIP URI) or a number in the PSTN (we refer to this information as *Recipient ID*).

The next step for the Voicemail Sender is to identify the destination network, i.e., the domain name or the IP of the gateway that serves the recipient. For achieving this, the Voicemail Sender utilizes the DUNDi protocol and more precisely its lookup operation. If the lookup provides the destination domain, the message is sent directly towards that domain; on the other hand, if the lookup operation returns no results, the voice message is blindly forwarded to the DTN ↔ Internet Gateway. Considering that the DTN ↔ Internet Gateway has a permanent connection to the Internet, it will have more up-to-date DUNDi information than the Voicemail Sender (that resides in DTN region).

Extending the above functionality, the DTN Voicemail station and namely its IP-PBX component can act as a repository for incoming voice messages (i.e. for messages that arrive from other DTN regions or the Internet).

B. DTN-Internet Gateway

The DTN-Internet gateway is a key component that serves two purposes: the first is to provide interoperability between the Delay Tolerant Network and the Internet, and the second is to locate destination users and forward received voicemails to them. To serve the abovementioned purpose the DTN ↔ Internet Gateway processes bundles which were sent to it by a DTN VM station (namely the Voicemail sender as described above). If the Voicemail sender could not identify the destination domain, the DUNDi protocol is used again for this purpose. Once the final destination is known, the bundle is converted to TCP/IP packets which are sent over the Internet towards their final destination. The DTN-Internet gateway also maintains a list of potential gateways for PSTN termination; one of these will be used to deliver the message when the recipient ID is a PSTN number.

C. Regional Voicemail Station

At the receiving end of a voice message, a set of components make up the Regional Voicemail station (RVM). The first point of contact is the Voicemail Receiver, the entity that receives the voice messages and starts the final delivery process by notifying the associated IP-PBX. According to the recipient-ID a number of actions are possible:

a) the recipient is a user of the local IP-PBX; a call is automatically placed to that user's extension and once it is answered the message is played back. If there is no answer, a voicemail is created (using a special script on the PBX) and deposited in the user's voicemail inbox.

b) the recipient is a PSTN number; a call is automatically placed to that number via the gateway connected to the local IP-PBX. Once the call is answered the voice message is played back. If there is no answer the call is re-tried after some time.

IV. CURRENT STATUS AND FUTURE WORK

Due to its delay tolerance, voice messaging is perfectly suitable for networks with intermittent connectivity. In this paper, we proposed a system capable of delivering voicemails that originate from users residing in areas with intermittent connectivity, to users in the outside world through the utilization of store and forward techniques offered by DTNs.

Our system provides interconnection to the PSTN which is a distinct advantage for the specific application we have in mind. Moreover, it is based on free PBX software and can use standard SIP-compliant soft phones or IP phones without the need for any special client software; user transparency and component interoperability are crucial features.

The proposed architecture is currently being implemented to offer the functionality described above (i.e. delivery of voice messages that originate from users inside the DTN regions). Once the initial implementation stage is complete, we plan to add functionality for message acknowledgement that can also form the basis for the delivery of incoming messages i.e. messages that originate from the PSTN and are destined for users in DTN regions. To facilitate such functionality we will have to add extensions to the IP-PBX. Moreover, to improve bundle delivery we plan to define an application layer protocol that allows PBXs to exchange information about message delivery and PSTN gateway availability.

REFERENCES

[1] Fall, K.: "A Delay-Tolerant Network Architecture for Challenged Internets", in Proc. Applications, Technologies, Architectures, and Protocols for Computer Communications, Karlsruhe, Germany 2003

[2] "Delay-Tolerant Networking Architecture", RFC 4838, 2007

[3] "Bundle Protocol Specification", RFC 5050, 2007

[4] "Delay Tolerant Networking TCP Convergence Layer Protocol", Internet Engineering Task Force Draft, 2008

[5] Wood, L., Eddy, W.M., Ivancic, W., McKim, J., and Jackson, C., "Saratoga: a Delay-Tolerant Networking Convergence Layer with Efficient Link Utilization", in Proc. International Workshop on Satellite and Space Communications, September 2007, pp. 168-172

[6] Demmer, M., and Fall, K., "The Design and Implementation of a Session Layer for Delay-Tolerant Networks", Computer Communications Journal, Vol. 32, Issue 16, pp. 1724-1730, 2009

[7] Lu X., and Hui P., "An Energy-Efficient n-Epidemic Routing Protocol for Delay Tolerant Networks", in Proc. 5th International Conference on Networking, Architecture, and Storage, Macau, China, 2010, pp.341-347

[8] Ramanathan, R., Hansen, R., Basu, P., Rosales-Hain, R., and Krishnan, R., "Prioritized Epidemic Routing for Opportunistic Networks", in Proc. 1st International MobiSys Workshop on Mobile Opportunistic Networking, San Juan, Puerto Rico, 2007

[9] Lindgren, A., Doria, A., and Schel, O., "Probabilistic Routing in Intermittently Connected Networks", SIGMOBILE Mobile Computing and Communications Review Vol. 7, Issue 3, pp. 19-20, July 2003

[10] Ott, J., and Pitk, M.J., "DTN-based Content Storage and Retrieval", in Proc. IEEE WoWMoM Workshop on Autonomic and Opportunistic Communications (AOC), Helsinki, Finland, June 2007, pp. 1-7

[11] Jones, P.C.E., Li, L., and Ward, A.S.P., "Practical Routing in Delay-Tolerant Networks", in Proc. ACM SIGCOMM Workshop on Delay-Tolerant Networking, Philadelphia, Pennsylvania, USA 2005, pp. 237-243

[12] Lindgren, A., and Phanse, K.S., "Evaluation of Queueing Policies and Forwarding Strategies for Routing in Intermittently Connected Networks", in Proc. 1st International Conference on Communication System Software and Middleware, Delhi, India 2006

[13] Jain, S., Fall, K., and Patra, R., "Routing in a Delay Tolerant Network", in Proc. Applications, Technologies, Architectures, and Protocols for Computer Communications, Portland, Oregon, USA, 2004, pp. 145-158

[14] Burgess, J., Gallagher, B., Jensen, B., and Levine, B.N., "MaxProp: Routing for Vehicle-Based Disruption-Tolerant Networks", in Proc. *25th* IEEE International Conference on Computer Communications, Barcelona, Spain, 2006, pp. 1-11

[15] "Use of the Delay-Tolerant Networking Bundle Protocol From Space", NASA Technical Memorandum TM-2009-215582 Report, 2009

[16] Lindgren, A., Doria, A., Lindblom, J., and Ek, M., "Networking in the Land of Northern Lights: Two Years of Experiences from DTN System Deployments", in Proc. ACM Workshop on Wireless Networks and Systems for Developing Regions, San Francisco, California, USA, 2008, pp. 1-8

[17] Udén, M., Grøttum, K.J., and Sjursen, S., "Realizing Delay Tolerant Networking as Access Enabler: Services Arising in New Realms and the Driving Applications", in Proc. eChalenge, Warsaw Poland 2010, pp. 1-7

[18] Seth, A., Kroeker, D., Zaharia, M., Guo, S., and Keshav, S., "Low-cost Communication for Rural Internet Kiosks Using Mechanical Backhaul", in Proc. 12th Annual International Conference on Mobile Computing and Networking, Los Angeles, CA, USA, 2006, pp. 334-345

[19] J.Scholl, L.Lambrinos and A.Lindgren, "Rural Telemedicine Networks using Store-and-forward VoIP and DTN", in Proc. Medical Informatics Europe, Sarajevo, Serbia, 2009

[20] Hull, B., Bychkovsky, V., Zhang, Y., Chen, K., Goraczko, M., Miu, A., Shih, E., Balakrishnan, H., and Madden, S., "CarTel: A Distributed Mobile Sensor Computing System", in Proc. 4th International Conference on Embedded Networked Sensor Systems, Boulder, Colorado, USA 2006, pp. 125-138

[21] Hui, P., Crowcroft, J., and Yonek, E., "Bubble Rap: Social-based Forwarding in Delay Tolerant Networks", in Proc. 9th ACM International Symposium on Mobile Ad-Hoc Networking and Computing, Hong Kong, China, 2008, pp. 241-250

[22] Hyyrylinen, T., Teemu, K., Luo, C., Jaspertas, V., Karvo, J., and Ott, J., "Opportunistic Email Distribution and Access in Challenged Heterogeneous Environments", in Proc. 2nd ACM Workshop on Challenged Networks, Montreal, Quebec, Canada, 200, pp. 97-100

[23] Islam, M.T., "DT-Talkie: Interactive Voice Messaging for Heterogeneous Groups in Delay-Tolerant Networks", in Proc. 6th IEEE Conference on Consumer Communications and Networking, Las Vegas, USA, 2009, pp. 21-22

[24] "SIP: Session Initiation Protocol", RFC 3261, 2002

[25] Asterisk - The Open Source Telephony Project: http://www.asterisk.org/, accessed April 21st 2011

[26] 'Distributed Universal Number Discovery (DUNDi) (Internet Draft)', 2004