SUCCESS STORY FOR R&D WORK ON VOIP

BACKGROUND

VoIP, the next big technology wave, is being adopted rapidly by companies in the communications industry. With multiple vendors introducing various VoIP services, it is imperative that companies have product differentiation to achieve global leadership. The client is an Atlanta-based consumer VoIP service provider enabling free conversations over the Internet using Two-Way Radios. Realizing the immense market potential for enabling two-way-radio (walkie-talkie) users to leave their computers and roam to the maximum range of the radios while talking for free over the Internet to anyone else on the peer to peer (P2P) network, Enosis was instrumental to support the client and initiate the R&D work for developing applications for connecting almost any Family Radio Service (FRS), General Mobile Radio Service (GMRS), or business two-way radio to a computer and then transmitting the user's voice over the Internet via a Voice over IP (VoIP) network in real time.

THE SOFTWARE

The system consists of two parts: the device and the software.

The 2Way device plugs into a standard USB jack. Then, using the cable provided with the device, user can connect a radio to the device. The 2Way software works like most instant messaging programs that feature a list of contacts. Making a call is as simple as speaking the words "Call Robert". The software uses voice recognition to understand your commands and either make or hang up calls with your contacts. Adding contacts to calling list is simple since all you need to know is your intended contact's email address and the 2Way application does the rest. Once you have created a list of contacts; you are free to pick up your second radio, move away from the computer & talk for free.

With long distance two-way radios you are able to get miles away from your computer and still communicate for free.

The 2Way software configures itself to work automatically through the toughest networks, NAT, firewalls, proxies, and UPnP routers enabling the user to make and receive Internet calls on the existing home phone for free. Another advantage of the system is that call receivers do not need a two-way radio and the 2Way device. They can download the 2Way software instead and connect to the network. It is also possible to call regular phone numbers and enjoy great voice quality at very low rates.

MAJOR CHALLENGES

- Preventing undue exhaustion of internet bandwidth
- Ensuring security of voice and data packets while passing through public network
- Establishing peer-to-peer connections from behind a restrictive firewall
- Providing smooth voice communication during unstable voice data transmission
- Eradicating the need for manual network configuration setting
- Providing support for voice command
- Facilitating natural user experience by supporting USB phone
- Ensuring interoperability by allowing other software to control the Two-way application

Software Phone Screenshot

ENOSIS SOLUTIONS' APPROACH

Due to the multi-dimensional expertise, VoIP domain knowledge and innovative approach exhibited earlier, the client chose Enosis to perform the R&D work for its VoIP product suite. Enosis allocated a dedicated R&D team to address the complexities by adopting a product R&D lifecycle approach. The steps in developing this application are delineated below:

Conceptualization: In the conceptualization stage, the R&D team created a prototype of the possible solution.

Design and Implementation: A unique combination of server-based and peer-to-peer voice/data communication has been developed for preventing undue exhaustion of internet bandwidth. The 2Way software has an intuitive contact list interface like many instant messenger programs so the user can call online friends in minutes.

Communication with server is encrypted using 128 bit RSA RC4 encryption. Microsoft CAPICOM technology has been used for the encryption.

Intuitive Contact List Interface Screenshot

Peer-to-peer communication is encrypted using blowfish algorithm to ensure voice-data is encrypted as it passes through the public network. Messages are sent in XML format. Communication with server is done via HTTP as this requires no firewall settings. The application automatically detects a user's proxy settings and supports HTTP, HTTPS, SOCKS 4 and SOCK5.

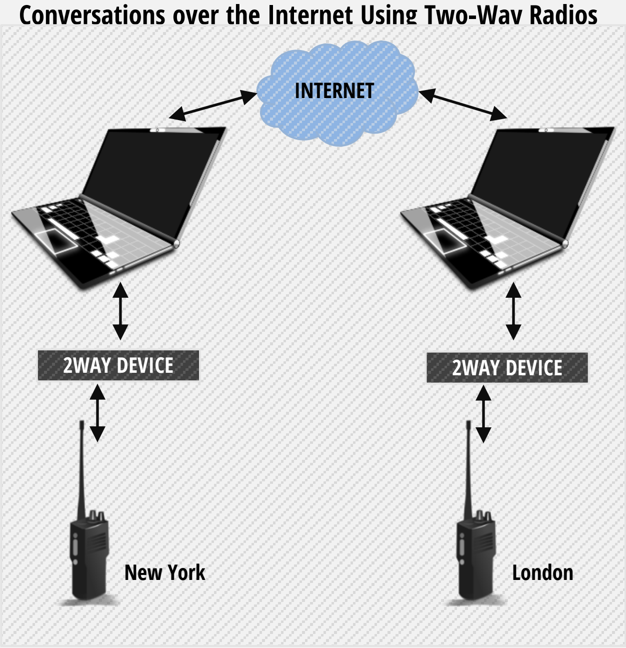
It is difficult to make peer-to-peer connections from behind a restrictive firewall. We used HTTP tunneling for these users. Here all potential users become tunneling servers (user is given choice to turn off tunneling feature unlike other software). Consequently a user behind a restrictive firewall will connect to multiple tunneling servers to communicate with their contacts (this is users helping each other).

The application design supports server migration - this means if the tunneling server, a user is connected to, becomes offline or too slow for data transmission, then it will automatically search and connect to the best and geographically nearest tunneling server. For ensuring smooth voice communication, the client establishes multiple p2p connections between users. It closes connections that become too slow for voice data transmissions and establishes new ones as necessary. This is done by automatically checking connection speed at regular intervals.

One-to-one voice and text chat, as well as voice conference between multiple users are supported while allowing users to have separate conversations with two or more contacts. The application allows users to put someone in the conversation "on hold" so that they do not hear what the rest of the people in the conversation are saying. It permits easy switching between the multiple conversations/ conferences a user is having. We used DirectPlay for voice communication.

The necessary tests to determine the firewall, proxy, and other network settings of a user are performed automatically, fully eradicating the need for manual configuration. It automatically decides whether a user can be a "fire hydrant", needs http tunneling or needs to go via a proxy etc. Here WinHttp is used for proxy detection.

Conversations over the Internet Using Two-Way Radios



ENOSIS SOLUTIONS' APPROACH

The service to let users make calls to regular phones (PSTN calls) is available using SIP protocol. On the server end it uses a powerful and scalable combination of SER and Asterisk servers. For a SIP client the application uses Microsoft RTC. Both the server and client are configurable so that any voice codec can be used. Currently G.723 is used for the voice codec. The service is setup as a prepaid service. The accounting is done by asterisk to calculate the balance for users. The database structure allows multiple packages (such as gold, silver, bronze etc.) for users to purchase, each package having different rates for countries. Additional features include pulse setting (currently set to 60 seconds), reseller options, multiple provider settings etc.

The 2Way application supports speech recognition allowing walkie-talkie users to make calls and avail useful features even when they are miles away from their PC. Microsoft SAPI is used for voice recognition and text-to-speech. For operating systems that do not support multiple capture buffers (like windows 98), it uses a "lower filter" driver to capture sound which is implemented using Windows DDK.

The privilege of using USB phones instead of headphones is provider for more natural user experience as users feel more comfortable using a telephone set rather than headphones. Currently in Eutectics phones and customized USB devices, functionalities like ring tones, DTMF detection, flash-detection, etc. have been implemented. The same application communicates with the customized 2Way USB device to interface with walkie-talkies.

DirectShow has been used to implement Voicemail and Webcam feature. User gets the privilege of choosing both the audio voice and video codec for voicemail and video recording respectively and playback at the destination. The application allows users to remotely control another computer. This has been implemented using VNC.

For ensuring interoperability, SDK has been developed to permit third-party software to control this application by allowing such third-party software to send commands and receive status message. This third-party control is implemented using Shared Memory.

Testing: In the testing phase, defects that were identified through Early Field Trials (EFT) were addressed. The test plans were written to cover all the test scenarios including feature validation and performance. Automation test scripts were developed using Windows ActiveX Component Architecture to facilitate reusability. These scripts were developed to test both deployment and validation of the system, reducing the manual testing time and resources.

Final Candidate for Shipment (FCS) and Sustenance: After the successful completion of EFT, the solution was packaged and released in over 10 countries across the North America, European and Asia Pacific markets.

Product R&D Lifecycle

REQUIREMENT MANAGEMENT	▪ FUNCTIONAL AND TECHNICAL REQUIREMENTS GATHERING ▪ PROJECT PLAN AND STANDARD DEFINITION, STORYBOARD AND GUI PROTOTYPING
DESIGN	▪ DEFINE TECHNICAL ARCHITECTURE, OBJECT MODELS ▪ DESIGN REVIEW AND SIGN-OFF
BUILD	▪ CODING AND UNIT TESTING ▪ CODE REVIEW FOR SANITY CHECK, STANDARD COMPLIANCE
TESTING	▪ INTEGRATION TESTING: COMPLETE TESTING FOR ALL TECHNICAL COMPONENTS ▪ SYSTEM TESTING: COMPLETE TESTING AGAINST REQUIREMENTS
UAT & DEPLOYMENT	▪ USER ACCEPTANCE TESTING ▪ ROLLOUT
POST IMPLEMENTATION SUPPORT	▪ SUSTENANCE-FIXING DEFECTS REPORTED DURING POST IMPLEMENTATION SUPPORT PERIOD

BENEFITS

- The software phone has an intuitive contact list interface like many instant messenger programs
- Facility to buy calling credits and make international calls at most economic rates with great voice quality
- Connecting two-way radios allowing "push to talk" across the internet and allowing connection between two remote sites that use radios without the need for repeaters
- The time taken to develop and deploy the solution was reduced by 30% and the cost was also reduced proportionally
- Client's association with Enosis enabled it to file a US patent for its innovation in VoIP domain

TOOLS AND TECHNOLOGIES

- **Microsoft Visual C++**
- **Microsoft DirectX**
- **Microsoft RTC**
- **Microsoft SAPI**
- **Microsoft Windows DDK**
- **Microsoft CAPICOM**
- **SIP Express Router**
- **Asterisk**
- **Microsoft SQL Server**
- **ASP**
- **VNC**

