Novel Architecture for Routing Packetized Voice Over Existing Internet Infrastructure Without Using the Internet Protocol

Kevork R. Piloyan,[†] *and Vahé Nerguizian*[††]*,*

[†] *Memotec Inc., Montreal, Canada* [††]*École de technologie supérieure, Montreal, Canada*

Summary
This article addresses a novel method to route Voice over the internet without using the Internet Protocol (IP). It talks about the problems of the current methodology to transfer Voice or Video traffic over IP (VoIP), and proposes a solution to these problems. This proposed solution consists of a new protocol that routes Voice and Video traffic over the Internet infrastructure. This novel protocol runs in parallel and independently from the Internet data routing protocols. Then it gives more detailed design description of this new protocol, and discusses different implementation methods. This is the first time a document on this technology has been written.

Key words:
Voice over IP, Internet Protocol, Routing, Bandwidth.

Current Technology

Voice over the Internet Protocol (VoIP) is a standards based technology and a widely used method to transfer and route voice, video and multimedia traffic over the Internet. There are many standard bodies, like the Internet Engineering task Force (IETF) and the International Telecommunication Union (ITU), with standards that address VoIP.

The most widely used and the oldest technology for VoIP is the ITU based H.323 [1] standard. H.323 consists of many sub-standards and has much evolved during many years since its conception in mid-1990's to allow seamless migration from the Plain Old Telephone Service [2] (POTS) to the Internet telephony using IP. With its supplementary services, H.450.1 [2], it could provide a similar service to that of POTS or ISDN [4] based switched telephony networks. Many vendors provide solutions to seamlessly integrate POTS and ISDN based networks with VoIP. These systems are called VoIP gateways.

A similar technology defined by the IETF standard RFC3261 [5], namely the Session Initiation Protocol (SIP), is gaining ground by popularity and becoming the de facto VoIP standard. This is due to its simplicity in integrating the current telephone system's signaling with that of SIP.

Manuscript received July 5, 2006.
Manuscript revised July 25, 2006.

This standard has been inspired by the current switched telephony network.

2. Actual Problems

The following sections describe some of the major problems associated with the current VoIP technologies.

2.1 Bandwidth utilization issues

The bandwidth calculations provided in Tables 1 and 2 (Appendix) assume two types of widely used and typical voice coder/decoders (vocoders), G.711 [6] and G.729 [7], which use 64Kbps and 8Kbps compressed voice packets respectively.

Table 1: Bandwidth utilization with G.711

G.711	***10 msec. Sample***[1]	***30 msec. Sample***[2]
Voice payload packet size in bytes[3]	80 bytes	240 bytes
Bandwidth without IP Headers[4]	64 Kbps	64 Kbps
Bandwidth with IP Headers[5]	96 Kbps	74.67 Kbps

[1] Represents the period of 10 milliseconds for a voice sample.
[2] Represents the period of 30 milliseconds for a voice sample.
[3] Represents the number of bytes of voice payload for the specified time sample in bytes.
[4] The bandwidth required to pass a voice packet sample on a link transfer medium.
[5] The bandwidth required to pass a voice packet sample over the Internet. The calculations are done on the basis of IP Header size (20 bytes), UDP Header size (8 bytes) and RTP Header size (12 bytes).

G.711	*10 msec. Sample[1]*	*30 msec. Sample[2]*
% increase in bandwidth[1]	50%	16.67%

Table 2: Bandwidth utilization with G.729

G.729	*10 msec. Sample[1]*	*30 msec. Sample[2]*
Voice payload packet size in bytes[3]	10 bytes	30 bytes
Bandwidth without IP Headers[4]	8 Kbps	8 Kbps
Bandwidth with IP Headers[5]	40 Kbps	18.67 Kbps
% increase in bandwidth[6]	400%	133%

As the calculations indicate, the overhead to transfer voice packets over IP is big. In order to keep the overhead low, the voice sampling should be done at bigger time intervals which means that the voice will get delayed by at least as much as the sampling period, without taking into consideration the delays on the transport link and the network. Voice delay will become obvious after 250ms of delay.

As an example, consider a 1.544Mbps pipe to transport voice packets. Without the IP overhead, the pipe could carry 190 simultaneous sessions without saturation when G.729 is used or it can carry as much as 24 simultaneous voice sessions when regular G.711 is used. However, if the IP overhead is included and a sampling of 10ms of voice is used to form a packet, then the number of simultaneous sessions will be reduced to 38 for G.729 and 16 for G.711 when using the same 1.544Mbps pipe.

In this same scenario, an extra voice call over the saturated bandwidth will result in packet losses; thus affecting all other voice sessions. This will result in choppy voice.
If buffering is used, then instead of choppy voice, there will be voice delays, since voice packets will be buffered before being processed.

Consider again this same scenario, but this time other Internet data traffic is sent on the same pipe that shares the same bandwidth with voice packets. This will also result in choppy voices and delays.

[1] Represents the bandwidth increase when a voice packet sample is sent over the Internet.

In order to resolve these issues, the IETF has come up with multiple types of solutions. The widely used is the Quality of Service (QOS, Differentiated Services RFC2475) [8] and the Multi-Protocol Label Switching (MPLS, RFC3031) [9]. Since the Internet is a mixture of routers and switches from different vendors, unfortunately, supporting QOS or MPLS on all nodes is impossible or complex to manage.

2.2 Security issues

Another problem is the combined deployment of VoIP and Firewalls. Most users have to open up some IP (UDP [10] and TCP [11]) ports to let the VoIP traffic pass, hence opening up their networks for attacks. It's rare to find an ISP or a user with an Application Level Gateway (ALG) for VoIP that bypasses the Firewall security. A good Firewall with this service will be very costly.

2.3 Routing issues

The internet core consists of high capacity and high bandwidth frame or cell switches. The core is unaware of traffic types; however it provides services and provisions for priority traffic by tools called circuits. The core rarely routes IP traffic. The main routing is done at the edge where Internet Service Providers (ISPs) provide IP services to customers. This means that each ISP controls the service for traffic prioritization. Some ISPs do provide them and some do not. Since VoIP and regular Internet data utilize the same routing protocol and use the same transport media, it cannot be relied on all ISPs to provide the same service for voice traffic.

Considering all these problems and limitations for voice over IP, it is very unattractive to deploy VoIP commercially, using the IETF or the ITU standards.

2.4 Solution considerations

There is a need for a new technology that overcomes these problems and is simple to implement, deploy and integrate with the current installed infrastructure. This new technology should not require firmware upgrade of the network core.

The following problems have to be addressed by this new technology:

- Bandwidth.
- Delay; voice is delay sensitive
- Packet loss
- Bandwidth sharing with other types of traffic
- Quality of service
- Call setup delays
- Firewall/security bypass issues.

2.5 Proposed Solution

A new protocol is the subject of the rest of this article that overcomes all the problems listed in previous sections. This new protocol, namely Internet Voice Routing Protocol (IVRP), which is derived from the same concept of IP, will run in parallel and independently from IP on the same media.

Major bandwidth problems occur on the WAN links where many type of applications share the same limited bandwidth. In order to overcome the bandwidth limitations, the edge router (as said in the earlier section that most routing is done at the edge), has to differentiate between application types in order to prioritize the traffic. By differentiating the IVRP traffic from regular IP traffic at the lowest link protocol layer will enable routers or switches to identify the type of the traffic at an early stage and take action accordingly.

As IP is based on source and destination addresses to route packets, and a protocol number to identify the type of IP payload, IVRP will also be based on source and destination dial numbers to route voice traffic, and a protocol number to identify different types of voice payloads.

As IP runs over different links, like LAN (Ethernet, Wireless LAN) and WAN (Frame Relay, ATM), IVRP will also run over the same links. It will require assigning special identification labels with standards bodies.

3. Protocol Description

In the following section, IVRP protocol details are described.

IVRP is a protocol similar to IP that is designed to pass voice and telephony related traffic.

Every site that supports an application for at least one voice connection must hold an outgoing-calls routing table. This table contains as many entries as possible remote phone/FAX sets that can be reached within the network. Each table entry is indexed using the extension dial entered as the local number and a mask associated with that entry. Each table entry provides to the local voice application all the routing information required in order to reach a specific remote phone/FAX set.

These table entries like IP routing database are either entered manually by a technician or are discovered automatically by registering local extension to all its node's neighbors, similar to the Routing Internet protocol (RIP) or Open Shortest Path First (OSPF) protocol.

This section is further divided into two major parts. The first part describes the format of a regular voice packet, and the second part discusses the Voice Routing protocol format.

3.1 Voice Packet Format

The following is the general format of a regular voice payload:

Table 3: General format of a regular voice payload

IVRP Header	***Voice payload***
8-266 Bytes	***1-1000 Bytes***

The IVRP Header further consists of the following format:

Table 4: The Internet Voice Routing protocol Header format

Field	***Bytes***
Version	1
Hops	1
Type	1
Ddnl	1
Ddn	1-128
Sdnl	1
Sdn	1-128
Channel	1-4
Payload	Variable

Table 5: Version bit field description

Bits	***0***	***1***	***2***	***3***	***4***	***5***	***6***	***7***
	Ver			ch		t	res	

Version, is a byte whose first 3 high bits represent the version of the IVRP protocol format. The Version of this protocol is represented as b001.
Bit 3 and 4, "ch", represent the channel field length. b00 represents a 1 byte channel number; b01 represents 2 bytes; b10 represents 3 bytes and b11 represents 4 bytes channel number.
Bit, 5 "t", is set for truncking scenario where multiple channels destined to the same remote - or point-to-point connections - are concatenated into one packet to save bandwidth overhead.
Bit 6 and 7, are reserved for future use.
Hops, is the time-to-Live of the packet. This value is decremented by one at each node or hop the packet is routed or switched. The default hops is a network configurable value.

Type, is 1 byte and represents the type of payload. Possible types are signaling (1), silence (2), G.711 (3), G.723 (4), G.729 (5), video, and so on. This field is a subject to be controlled by a standards body.
Ddnl, is one byte and represents the length of destination dial number (ddn) in nibbles (4bits, or dial digits).
Ddn, is the destination dial number. This number represents the information on which the IVRP protocol will rely to route the packet. It does not represent the actual dial number of the destination telephone address. This field consists of 2 times 4 bit nibbles which represents a dial digit. Maximum number of dial digits on which routing is performed is 256 digits which corresponds to 128 bytes.
Sdnl, is one byte and represents the length of source dial number (ddn) in nibbles (bytes, or dial digits).
Sdn, is the source dial number. This number represents the information on which the IVRP protocol will rely to route the packet back to its source. It does not represent the actual dial number of the source telephone address. This field consists of 2 times 4 bit nibbles which represents a dial digit. Maximum number of dial digits on which routing is performed is 256 digits which corresponds to 128 bytes.
Channel, is the channel number assigned to the particular call session between the same peers. This field could be up to 4 bytes long depending on the high two bits of the options field.

If the network is set up as point-to-point and multiple voice calls are possible, then the trunking method can be used to save extra overhead. With this method, the same packet header is used for multiple voice payloads.

The following is the IVRP header and packet format for trunking application:

Table 6: IVRP header and packet format for trunking application

Field	***Bytes***
Version	1
Hops	1
Type	1
Ddnl	1
Ddn	1-128
Sdnl	1
Sdn	1-128
Number of channels	1
Channel 1	1-4
Payload 1	Var
Channel 2	1-4
Payload 2	Var
…	…
Channel n	1-4

A new field in the IVRP header is the Number of Channels which represents the number of channels that are concatenated in the packet after the IVRP header.

3.2 Voice Routing Packet Format

Voice routing is built on the same concept of Internet's RIP or OSPF.

Voice routing tables or database entries are kept and are used by the routers to route a packet. These routing tables are either entered and configured statically by a user or created dynamically by automatic registration of local dial numbers into the routing database and advertising these dynamic entries to neighbor routers by means of the Internet Voice Routing Protocol.

The IVRP is a protocol that shares and advertises local and remote Voice Routing Databases with immediate neighbors. It shares information such as dial numbers, network interface numbers, costs, and backup routes. IVRP should identify automatically duplicate routing entries and update the database entries with the latest advertisement. It should also identify more significant routes, and replace the less significant routes with more significant routes. The same should be done for less costly entries against costlier routes.

The following are information that is stored in a Voice Routing Database entry:

- Dial number including wild characters
- Length of dial number
- Interface to which the voice packet should be switched once the dial number is matched.
- Backup interface, if the primary interface is down
- Cost of the route.

The following is the format of the IVRP protocol packet:

Table 7: Internet Voice Routing protocol packet format

Field	***Number of records***	***Record 1***	***Record 2***	***…***	***Record n***
Bytes	1	Variable	Variable	…	Variable

Record, represents routing database entry advertisement information and its format is as follows:

Table 8: Record format for IVRP protocol packets

Record Field	***Size of dial Number***	***Dial Number***	***Cost***
Bytes	***1***	***Variable***	***Variable***

Route advertisement is performed at a pre-configured time interval or as new dial numbers are introduced on the network.
There is also the concept of default route. When a voice packet is to be transmitted and its route is not resolved, then the default route will be activated for that packet. Routing should be able to route from any transport media to any. As this protocol will run mostly on the edge, it must consider the routing between LAN protocols (Ethernet, Wireless media) and WAN protocols (Frame Relay, ATM and HDLC).

4. Implementation

As described above, Internet Voice Routing is a protocol that runs in parallel to the Internet Protocol (IP). This means that it can use the Internet backbone hardware to run. IVRP runs in a higher priority than the regular internet traffic, thus with more quality of service. Having to share bandwidth with regular IP traffic will not be a problem.

First let us calculate the bandwidth savings that a user will have versus the traditional VoIP topology using a typical configuration.

Assuming a three digit routing dials, and a maximum of 8 E1 (30 channels per E1 - 240 channels maximum) interfaces, the IVRP packet header will be 10 bytes. The breakdown is as follows:

- 1 byte for version
- 1 byte for hops
- 1 byte for type
- 1 byte for destination dial number length (value is 2)
- 2 bytes for the destination dial number (as 0XXX where X is a dial number)
- 1 byte for source dial number length (value is 2)
- 2 bytes for the source dial number (as 0XXX where X is a dial number)
- 1 byte of channel number (up to channel 240).

As seen from the two following tables, the bandwidth savings for IVRP is significant. This bandwidth savings become even more significant when this technology is used over low speed, high cost transport links, such as satellite or wireless.

[7] The bandwidth required passing a packetized voice sample over the Internet using the IVR format. The calculations are done on the basis of IVRP Header size of 10 bytes.
[8] Represents the bandwidth increase when a packetized voice sample is sent over the Internet.
[9] Represents bandwidth savings over the traditional VoIP protocols, calculated as such: Bandwidth with IP Headers - Bandwidth with IVRP Headers.

Table 9: Bandwidth Savings calculations for G.711

G.711	*10 msec. Sample*	*30 msec. Sample*
Voice payload packet size in bytes	*80 bytes*	*240 bytes*
Bandwidth without IVRP Headers	*64 Kbps*	*64 Kbps*
Bandwidth with IVRP Headers[7]	*72 Kbps*	*66.67 Kbps*
% increase in bandwidth[8]	*12.5%*	*4.17%*
Bandwidth Savings over VoIP[9]	*24 Kbps*	*8 Kbps*

Table 10: Bandwidth Savings calculations for G.729

G.729	*10 msec. Sample*	*30 msec. Sample*
Voice payload packet size in bytes	*10 bytes*	*30 bytes*
Bandwidth without IP Headers	*8 Kbps*	*8 Kbps*
Bandwidth with IVRP Headers[7]	*16 Kbps*	*10.67 Kbps*
% increase in bandwidth[8]	*100%*	*33%*
Bandwidth Savings over VoIP[9]	*24 Kbps*	*8 Kbps*

Development of such a protocol can be very tricky, as it should take into consideration its similarity with the IP protocol, yet its independent execution from IP must be forced. IVRP should be run in parallel to the IP protocols, and should be given higher priority to that of IP and its applications. This means its coexistence with other protocols running at the same network layer is a must.

As far as implementation of a network with IVRP protocol concerned, the dial numbers can get very tricky. The more digits the dial is based on, the bigger the packet, hence the more bandwidth it requires. The implementation should also be aware of different regions where dial numbers can differ in length. The implementation should restrict this complexity.

The implementation should also consider to further subdivide regions (or dial numbers) when the installation base gets large, instead of increasing the dial digits for routing. This means that the routing protocol should allow the definition of regions and not allow detailed routing information transfer between each node of all regions. Only dedicated border routers should exchange such information. When network gets bigger, the user can also combine (trunk) multiple channels into one packet for all point to point traffic. This will result in some bandwidth optimization.

Having IVRP a protocol independent and of higher priority from that of IP, will allow the user to overcome the problems of quality of service and hence delays. As soon as a router identifies an IVRP packet, it will route in priority.

The user will also overcome the problems of firewalls where IVRP will automatically bypass the firewall without jeopardizing network security since it is not based on any IP protocols.

One main point worth mentioning is the ability to reuse the same hardware existing on the network. It uses the same CPU processor, the same network adaptor cards or WAN serial cards. However, it could be problematic where proprietary hardware is used or application specific hardware like CPUs that run in conjunction with network processors designed for the IP protocol. In most cases, a simple Software or firmware upgrade is adequate.

As the Voice over IP protocols, IVRP also has its good and bad points.

The followings are some of the main good points of the IVRP protocol:

- IVRP uses the same hardware infrastructure as the Internet. This reduces the deployment costs
- Changes are only on the edge. There is no need to reconfigure or upgrade core equipment
- If routing dial numbers are kept small, the overhead will be minimal
- Use trunking in point to point nodes where there are a lot of simultaneous voice calls
- IVRP can inter-work with many transport layers such as Ethernet (LAN) and FR/ATM (WAN), since it is independent of the transport link layer.

Some of the major bad points of the IVRP protocol are the followings:

- Software or firmware upgrade is required for all equipments
- Interoperability with different vendors could become an issue if a standards body does not enforce protocol rules.

5. Conclusion

In this article we have discussed about a new protocol called the Internet Voice Routing (IVRP) protocol that overcomes most of the current Voice over IP standard protocols. Among those problems include bandwidth issues, quality of service, packet loss due to congestion and delay, and last but not least, firewall by-pass issues.

We have also defined an IVRP packet and discussed about its headers. Then we have addressed an automatic voice route learning mechanism, similar to that of the IP routing protocols, based on a dial number. And last we have discussed about implementation procedures and issues.

Many protocols, as such, has been defined, developed and implemented that overcomes the problems discussed in this article about Voice over IP protocol. However, none has really addressed and resolved all problems simultaneously with one protocol. IVRP addresses bandwidth savings, quality of service, packet loss due to congestion and delay, and last but not least, firewall by-pass issues - all, at the same time

References

[1] H.323, Packet-based multimedia communications systems, ITU-T, February 1998.
[2] POTS, Plain Old Telephone Services, is also called the Public Switched Telephone Network (PSTN).
[3] H.450.1, Generic functional protocol for the support of supplementary services in H.323, ITU-T, February 1998.
[4] ISDN, Integrated Services Digital Network, ITU Series I and Q recommendations.
[5] RFC 3261, Session Initiation Protocol, IETF standard, J. Rosenberg, H. Schulzrinne, G. Camarillo, A. Johnston, J. Peterson, R. Sparks, M. Handley, E. Schooler, June 2002.
[6] G.711, Pulse code modulation (PCM) of voice frequencies, ITU-T, November, 1988.
[7] G.729, Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear-prediction (CS-ACELP), ITU-T, March, 1996.
[8] RFC 2475, An Architecture for Differentiated Service, IETF Standard, S. Blake, D. Black, M. Carlson, E. Davies, Z. Wang, W. Weiss, December, 1998.
[9] RFC 3031, Multiprotocol Label Switching Architecture, IETF Standard, E. Rosen, A. Viswanathan, R. Callon, January, 2001.
[10] UDP, RFC 768, User Datagram Protocol, IETF Standard, J. Postel, August, 1980.
[11] TCP, RFC 793, Transmission Control Protocol, IETF Standard, J. Postel, September 1981.

Kevork R. Piloyan, B. Eng., MBA, is an executive who has been working for over 18 years in the Telecom industry in R&D and business development. He has worked in Software Development, Product and Project Management, and Business Development Management in various leading Canadian and US companies. He has held positions in both business as well as technical management.
Kevork R. Piloyan is an inventor, and an innovator. He has many technology inventions in data and voice networking, and has published journals and articles on them. He has spoken in leading technology conferences around the world, and has also taught in an institution of higher education. He has actively participated in several company acquisitions and mergers, and has taken active roles in business and technology management of these companies.

Vahé Nerguizian, B.Eng., M. Eng., received the B.Eng. and M.Eng. degrees in electrical engineering from École Polytechnique and McGill University in 1981 and 1983. He has over 14 years of industrial experience in RF, microwave and optical fields with Canadian Marconi Company (CMC) and Nortel Networks. For several years, he was also program manager in the hybrid component packaging department at CMC. Currently, he is an associate professor at École de Technologie Supérieure (ÉTS) at the department of Electrical Engineering - University of Québec in Montreal. He is a member of LACIME research group at ÉTS. His research interests are in Microwaves, MEMS for RF/Microwave and millimeter wave applications, MEMS for robotic applications and MEMS packaging in general. He has spoken in leading technology conferences around the world about
Professor Nerguizian is a member of the Institute of Electrical and Electronic Engineering (IEEE), the international Society for Optical Engineering (SPIE), CRIAQ (Consortium for Research and Innovation in Aerospace in Quebec) and the Order of Engineers of Quebec (OIQ).