VoIP: Voice Over Internet Protocol

Acronyms

VoIP

Voice over Internet Protocol

STUN

Standard Traversal of UDP NAT

SIP

Session Initiation Protocol

Benefits of using VoIP

VoIP has many features over the standard telephone system. The two primary reasons for a switch over to VoIP is lower cost and increased functionality.

Because standard telephone services today are controlled by corporations or government entities there is an increase in the cost to provide this service. VoIP seeks to resolve this issue because, in essence, each user of VoIP is their own service provider. Aside from the standard cost of Internet connections, making calls using VoIP is free.

VoIP also allows features which are either made more expensive by standard phone services or almost impossible. Some features of VoIP packages out there include easier methods of conferencing, blocking specific phone numbers, or automated replies. One such software package is Asterisk which can be found at http://www.asterisk.org/. Asterisk is open source software for the Linux platform.

NAT and VoIP

As covered previously in class NAT or Network Address Translation is a means by which multiple devices on a LAN with private IP addresses can share a single public IP address. NAT is used by firewalls and routers to translate between private and public IP's at the point where the LAN connects to the Internet.

One of the problems of NAT is that when a connection is initiated by a device on the Internet outside of the LAN there is no way to determine which device on the LAN the connection was meant for. The NAT router may support simple rules on what is known as a 'software DMZ' or port forwarding for services such as web and mail however these are both insufficient if there are multiple devices on the LAN to which a certain service may apply to.

Furthermore, the way conventional VoIP protocols are designed problems with the passing of VoIP traffic through NAT arises. Currently most conventional VoIP protocols only deal with the signaling of a connection whereas audio traffic is handled by another protocol. To make things worse the port on which the audio traffic operates is chosen completely at random. As a result the NAT router will be able to handle the signaling traffic, but it begins to fail when dealing with the audio traffic as it has no means of associating the resulting audio connection with the signaling traffic.

How to deal with the NAT and VoIP incapability

There are several ways to deal with this incapability. The most obvious of solutions is to avoid the problem with NAT to begin with. Ways to do this is to not use NAT and obtain public IP addresses, use IP tunneling, or use a NAT friendly VoIP protocol.

The other solution is to use one of the several workarounds available. Further information on these workarounds can be found at http://www.voip-info.org/wiki-NAT+and+VOIP.

Misc. Topics

Co-branching

Co-branching involves the broadcasting of a call to multiple IP addresses. In essence each user becomes their own phone provider. As a result a particular feature of co-branching is that a user can opt to have all their calls, for example, forwarded to their conference room while they are out of the office.

911 Service

Right now a user can not really use the 911 service using VoIP. The reason behind this is that the SIP server may have an idea of where the location of the telephone is, or if the phone is behind a media server they may know where media server is. However, if a person is on the road for example, then the SIP server may be located back at the individuals office and offer no insight as to the location of the cell phone. What would be needed is a way to map physical location to an IP address which is not possible for multiple reasons.

Signaling

Possibly one of the most important aspects of VoIP. There are currently three signaling standards.

H323

H323 was the first signaling protocol. It was developed by the International Telephony Union which is a subset of the United Nations who is also responsible for country codes. The webpage for this is www.itu.int.

SIP

Session Initiation Protocol which was created and backed by the IETF.

Skinny

Developed by CISCO.

Currently SIP seems to be carrying the most weight and as a result is what will be covered mostly.

SIP Continued

SIP has three primary modes of operation. It runs over UDP, TCP, and TCP/TLS. On UDP and TCP it runs on port 5060 and on TCP/TLS it runs on port 5061. It follows similar methods as those used in the HTTP protocol.

REGISTER method

Registers a phone with the SIP server. We can also register a phone to a name, not just a number. Register also allows authentication of a phone.

INVITE method

Uses the SIP server to invite a phone. Similar to a DNS look up in that if the invite succeeds the other phone starts ringing though there are still differences

CANCEL method

Cancel method is sent by the caller when an invite is generated. This is invoked if an individual hangs up before the connection is made.

RING method

Reply back to an invite. The RING method is an indication that the other phone is ringing.

BYE Method

Can be initiated either side and indicates the the connection is to be terminated.

ENUM

ENUM is a protocol maintained by ITU that seeks to find a way to map a telephone number to a URI, Uniform Resource Identifier. It uses the notion of E.164 numbers which can be found RFC 2916.

NAPTR: Naming Authority Pointer

The NAPTR record consists of a number of fields and supports several features. One such feature is that NATPR records are recursive in that they can call other NAPTR records.

E164 Types

e164.int: UN

e164.arpa: IETF

e164.info

e164.info was created by several phone companies who grew tired of the argument between the supporters of e164.int and e164.arpa.

Power over Ethernet

Power over Ethernet uses the 1,2,3, and 6 cables as standard. The remaining four cables are tied together in pairs, that is two pairs of two cables tied together. The result is the ability to deliver 48 V DC.

Media Stream

RTP

RTP is a UDP based protocol. It's RFC number is 1889. It's based to send the packet once and also have some overlap in the packets in order to attempt reconstruction if absolutely needed. However it's goal remains to deliver the packets as fast as possible.

1 Conclusion

The above notes are quite incomplete and for that I apologize. I have included a few URLs below that will further elaborate on the topics covered in class.

VoIP, General: http://www.voip-info.org/

E.164 Numbers and DNS: http://www.faqs.org/rfcs/rfc2916.html

RTP: A Transport Protocol for Real-Time Applications: http://www.faqs.org/rfcs/rfc1889.html