Voice Over IP - Stepping towards a converged network

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The traditional phone network has existed for over a century now, providing phone service to millions of people all over the world. This service has been characterized by ever-increasing quality and ever-decreasing cost. New service features, new methods of access, new protocols have been introduced. However, the network has become complex, the equipment expensive, and it takes a long time to add new features.

Over the past decade, many companies have turned their attention to providing voice calls over the Internet. Applications such as Toll Bypass, Alternate Long Distance, Voice Virtual Private Networks, IP Centrex, Local Calls, and 800-Number Service have been developed and deployed. These applications were mostly proprietary, and for the end customer to use these applications, it was necessary for him to sign up for service with one Service Provider, who in turn could obtain his equipment from one Equipment provider.

The last two years have seen a tremendous surge in internet usage, and has brought voice to the fore-front in internet applications – free internet phone calls, voice capability in instant messages, broadcast video and audio (recorder and real-time), voice enabled applications for customer service and e-commerce – have all been developed and deployed by end customers, internet portals, internet service providers and enterprise customers.

With a market estimated at over $10 billion in 3 years, this has provided a surge on interest among both equipment providers and service providers. However, in order for IP Telephony to become a mainstream application three major steps are required –

a) Moving away from proprietary solutions to standards based solutions.
b) Providing service quality that is comparable to the PSTN.
c) Providing transparent interoperability of the IP and PSTN networks.

Requirements for IP Gateways.

In the move from proprietary products to standards based products, many of the "features" provided by earlier applications have now become "requirements" for new gateways. Some of these features/requirements are

1. Mobility – move with the user, provide same user interface everywhere.
2. Universally accessible service – available to every user, everywhere,
3. Interoperability across protocols, vendors, service providers, and across national boundaries.
4. Reliability – the network should be robust, distributed, and fault tolerant.
5. Scalability – scale from small number of users to large enterprise networks
6. Security – authentication, encryption.
7. Subset of standard applications - Automated call handling, calling card services, Internet Voice and Fax service, Voice VPN, video conferencing, Voice enabled PC applications.

Matching the PSTN.

Before the IP network can gain widespread deployment, it must be able to match, if not exceed, the quality of the PSTN.

The IP network must be able to provide real-time audio, with acceptable voice quality. Two standards have been defined for Subjective and Objective measurements of voice quality. Features of voice quality are delay, packet loss, jitter, echo cancellation, clarity, and silence suppression.
The acceptable delay for voice lies in the range of 0 – 100-ms. Adaptive jitter management and comfort noise generation are some of the methods used to compensate for jitter and silence.
The IP Network should duplicate the existing, accepted features of the phone network – caller-id, numbering, routing.

The IP Network should provide some tangible benefits over the phone network.

<u>Transparent Interoperability.</u>

The IP Telephony Gateway must work transparently with the PSTN network. It must work across different national boundaries, different service provider networks, different vendor equipment and different protocols.

<u>Gateway Decomposition.</u>

An IP gateway can be divided into three components – media gateway (MG or MGW), signaling gateway (SG or SGW), and media gateway controller (MGC).

The ***signaling gateway*** is responsible for signaling between end-users on either network.
On the PSTN side, SS7 or ISDN (T1/E1-BRI/PRI) is used, which is then translated to an IP signaling protocol such as SIP or H.323, and transported across the IP network.
SAP (Session Announcement protocol) is used to announce the session.
SDP (Session Description Protocol) is used to describe the call (or session).

Once a call is setup, the ***media gateway*** is responsible for transfer of the data/video/audio streams.
On the PSTN side, media transport is by PCM encoded data on TDM streams.
For the IP network, different encoding formats (codecs) are used, to transfer data over UDP streams. The protocol uses is RTP. Other protocols are RTSP and RTCP.
The media gateway also handles echo cancellation, security and other end-user features.

The ***Media Gateway Controller*** is used to control one or more Media Gateways.
Megaco (H.248) and MGCP provide a mechanism for controlling multiple gateways. Some policies can be default call handling, default user profiles, codec selection policies, and QoS settings for customers.

<u>VoIP Protocols.</u>

VoIP uses a number of protocols, some of which are mentioned above.

As far back as 1994, the ITU introduced its H.323 family of protocols, to provide multimedia capability over the Internet. Many vendors have developed and deployed these solutions. In parallel, the IETF introduced many protocols used for IP Telephony – RTP, RTSP, RTCP, Megaco, SIP, and SDP. These protocols provide the foundation for standards based IP Telephony.

H.323 (H.235/H.245/H.225 (Q.931 and RAS) uses concepts such as terminal, gateway, gatekeeper, and media processor. It uses TCP as the underlying protocol for setting up connections.
MGCP uses concepts such as endpoints, connections and calls. It works over TCP or UDP.
Megaco uses concepts like terminations and endpoints. It uses optional packages to add capability to gateways.

SIP (RFC 2205) evolved as an alternate to H.323. It works over TCP and UDP. Provides a centralized or distributed call model, and re-uses Internet addressing schemes such as email address, IP address, url. H.245 uses TCP. If a connection is lost, the call is lost. H.245 declares a call as connected as soon as the call setup is complete, without checking the underlying media path. SIP verifies the media path is ok.

RTP or Real Time Protocol is the standard protocol used for media transfer. For PSTN, DTMF signaling is used. At the IP Gateway, these digits are collected, analyzed and sent over RTP. RTP profiles are defined, for carrying DTMF Digits, Tones, multicast, users behind firewalls. RTP header compression can be used to increase bandwidth.
RTSP or Real Time Streaming Protocol is used for broadcast (multicast) video and audio.
RTCP or Real Time Control Protocol is used to monitor and collect statistics on the performance of the Gateway. This protocol is not really useful at present, since in order for the measurements to be meaningful, it has to be supported by every gateway in the path between two end-users.

Codecs are used to encode and compress data for transport across the Internet. Common codecs used are G.711. G.723 and G.729. Most codecs use 8 KHz sampling and a default packet rate of 20 ms. Which codec to use, is negotiated at the time of setting up a call, and is determined by end-user capability, bandwidth, and codec policies imposed by the service provided. It is sometimes possible to change codecs during a call, (e.g. with SIP). G.711 is commonly used. However, use of G.711 over RTP wastes bandwidth. Hence G.729 is the preferred codec.

No discussion of VoIP can be complete without mentioning ATM and packet cable.

ATM is already used extensively, to provide the backbone of both the telecom and datacom networks.
ATM can carry IP in many forms. In addition, many of the IP Telephony protocols mentioned above, can work with ATM or over ATM. Call signaling can be handled by ATM itself, or the call signaling protocol of choice can be carried over ATM. Finally, since ATM can provide controlled delay, there is no need for echo cancellation or compression, and hence no need for Codecs.

Finally, the cable companies have come up with the ***packet cable*** standard, to provide voice capability and PSTN connectivity over cable.

VoIP Issues.

There are a number of issues with VoIP today. Different vendors are providing different solutions.

Routing – Routing calls worldwide necessitates the building up of huge routing databases of users, email addresses and IP addresses. This can quickly become unmanageable. One solution is to use static routing tables.

Usage records can be used for usage analysis and troubleshooting, but not for billing, unless security is guaranteed.

Security. There must be user authentication, encryption of data, especially for e-commerce application. This is provided by a variety of means – firewall, encryption software or hardware, accept calls only from known gateways.

Quality of Service –Guaranteed bandwidth for a user, codec compression vs. excess bandwidth, prioritized messages, statistics, monitoring and control.
There are no guaranteed solutions, however, vendors provide some mechanisms for each of these issues – use RTCP for control, use the Expedited forwarding bit for prioritized messages, provision for higher bandwidth (if bandwidth is cheaper) or use codecs and compression if bandwidth is expensive.

Codec licensing – Since royalties need to be paid for most codecs, companies need to track codec usage, both for payment of royalties, and in turn, for billing.
One solution is to use a license server, and allow the user to select codecs or bandwidth.

Using an IP Telephony gateway.

An IPT gateway using SIP is used to demonstrate call flows.
Call control gives the user full control of a call. With SIP, both distributed and centralized call control can be used. Call control using a distributed model is shown.
SIP uses client-server architecture. A user agent client originates a call. The server responds to the user agent.

Three different types of servers are defined – a proxy server, which can accept calls on behalf of an end user, and forward them to the user, wherever he may be, a re-direct server, which can find out where a user is and sends back the information, so that the use can place and complete a call, and a registration server or location server, which keeps track of where users are.

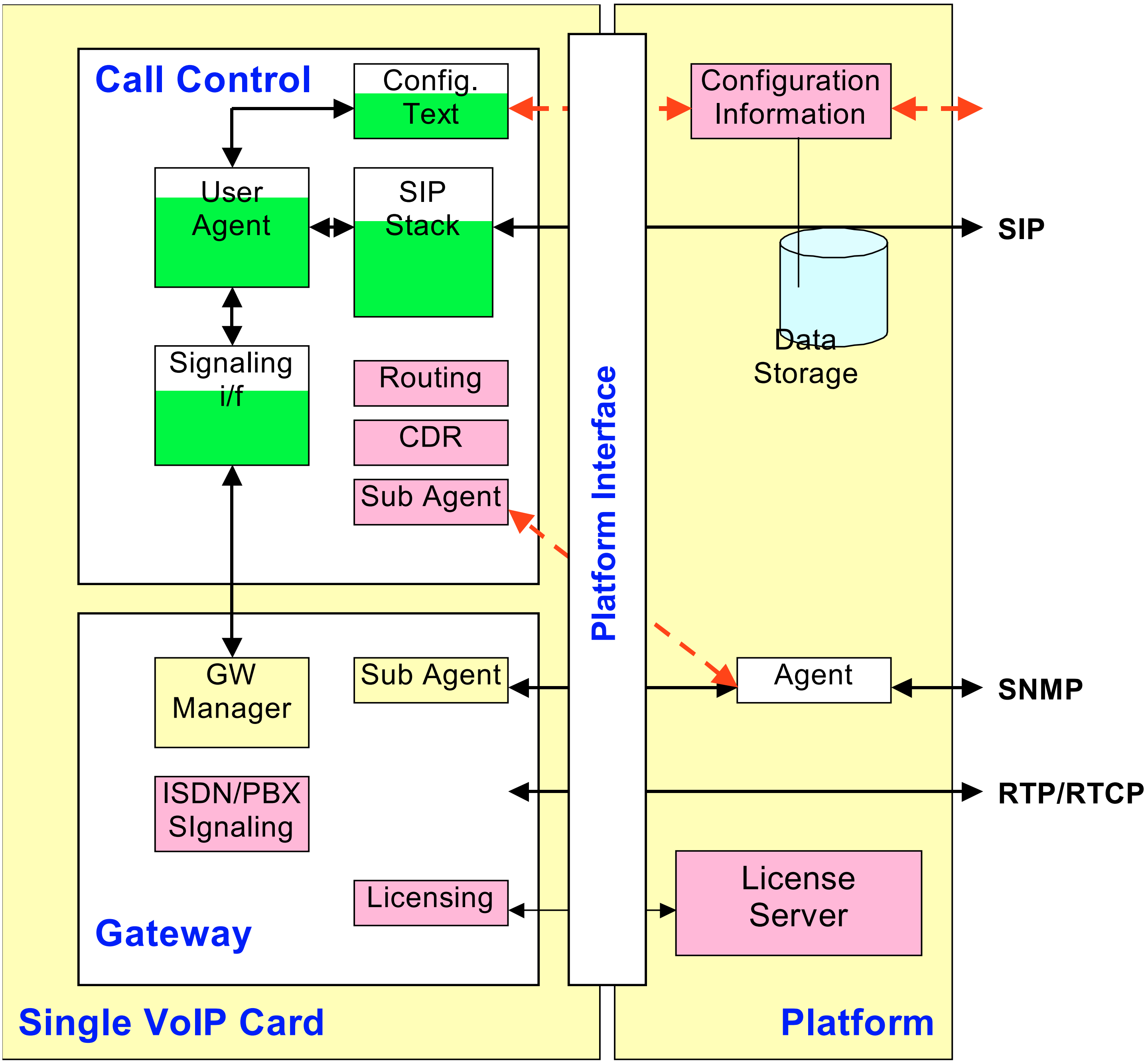
When a user places a call, using an email address, INVITE sip:cm@nokia.com
the address gets resolved to a location – user@host. (cm@swlab1.nokia.com)

The user must be registered with the registration server, with his new address (or can set uo a floow me service). – REGISTER cm@ieee-kerala.org.

If a server is unable to resolve an address, it will contact a registration server (location server) to discover where the user is. After determining the user's location, a re-direct server informs the user (302 - MOVED sip:cm@ieee-kerala.org.), so that the user manually competes the call. A proxy server would complete the call with the new address (INVITE sip:cm@ieee-kerala.org. 200 OK.) If the location server is unable to resolve an address, a failure message is sent back to the caller's user agent client.

Conclusion.

VoIP is an important part of the communications network of the future. It has a broad range of applications and huge market potential.
RTP, SIP and Megaco are emerging as standard protocols for IPT Gateways.



Typical Gateway Architecture

Standards based VoIP products will soon replace portions of, or become part of, the PSTN network. Finally, there will be a converged network, which is a mix of PSTN and IP Gateways.

References.

•IETF: http://www.ietf.Org.
–Megaco, SIP, PSTN-internet interworking.
•Http://www.ietf.Org/html.Charters/iptel-charter.Html.
•Http://www.Ietf.Org/html.Charters/Megaco-charter.Html.

- Http://www.ietf.Org/html.Charters/sip-charter.Html.
- Http://www.ietf.Org/html.Charters/pint-charter.Html.
- Http://www.ietf.Org/html.Charters/avt-charter.Html.
- Http://www.ietf.Org/rfc.Html.
- Http://www.ietf.Org/id.Html.

- ITU: http://www.itu.Int.
    - H.323 - http://www.itu.int/itudoc/itu-t/rec/h/h323.html.
    - Q.861 – Perceptual Speech Quality Measurement.
- SS7 http://www.Webproforum.Com/ss7/index.Html.
- SMIL: Http://www.W3.Org/AudioVideo.
- Packet cable: http://www.Packetcable.Com.