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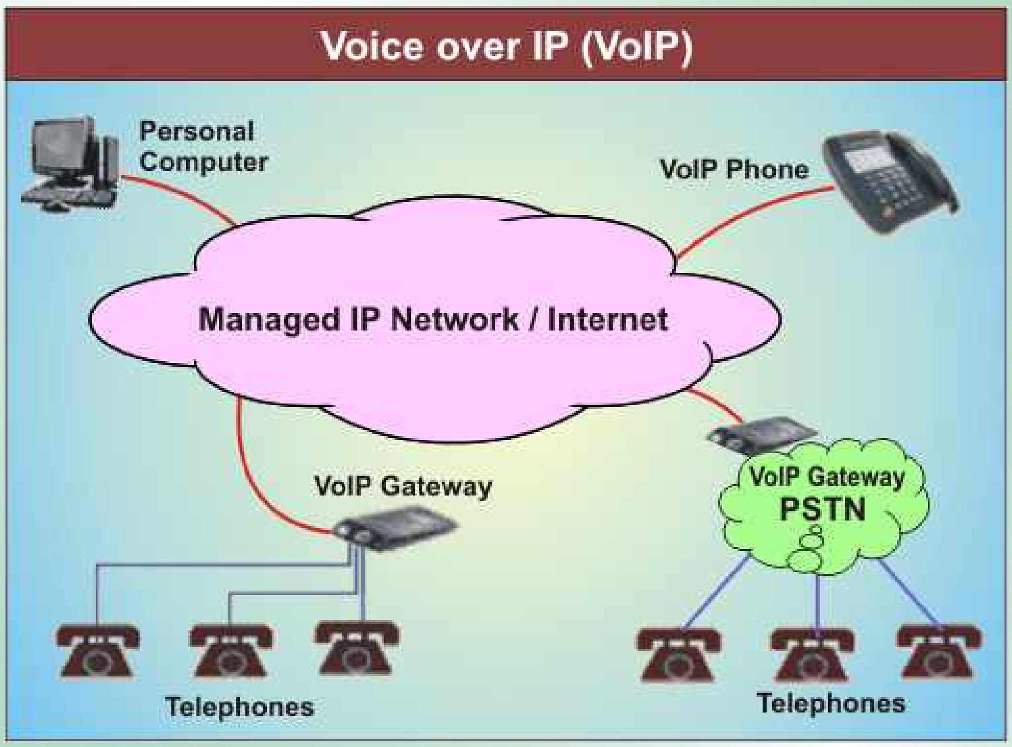



## IN THIS ISSUE

# नववर्ष के शुभ अवसर पर हार्दिक शुभकामनायें 2006

## Voice over IP (VoIP)

At its simplest, Voice over IP (VoIP) is the transport of voice using the Internet Protocol (IP). However, this broad term hides a multitude of deployments and functionality and it is useful to look, in more detail, at what VoIP is being used for, today. Currently the following types of VoIP applications are in use:

- Private users who are using voice over IP for end to end Voice calls over the public Internet (services like Skype, Vonage etc.). These users typically trade quality, features and reliability for the fact that the service is very low cost and are generally happy with the service. Although globally the numbers of users taking advantage of this technology is large, the density of such users is very low and when compared with the PSTN, the call volumes are negligible.
- Business users on private networks provided by telecom and datacom providers. These services offer relatively high quality and reliability and are feature rich but come at a price. When compared with the PSTN the call volumes supported by these services are small, however, such services are nonetheless commercially successful.
- IP trunking solutions used by long haul voice providers. Typically these offerings use private IP networks to connect islands of the PSTN. Customers access these services using traditional black phones, but the voice is carried over an IP network.

Although these voice over IP deployments have been successful and each will continue to have its place in the future, they have not yet faced the issue of how the wider PSTN could be migrated to an end-to-end voice over IP infrastructure. Providing a voice over IP solution that will scale to PSTN call volumes, offer PSTN call quality and equivalent services, as well as supporting new and innovative services is a significant challenge.

### Issues in a VoIP Network

There are several issues that need to be addressed in order to provide a toll-quality, PSTN equivalent end-to-end VoIP network. Some of them are discussed below:

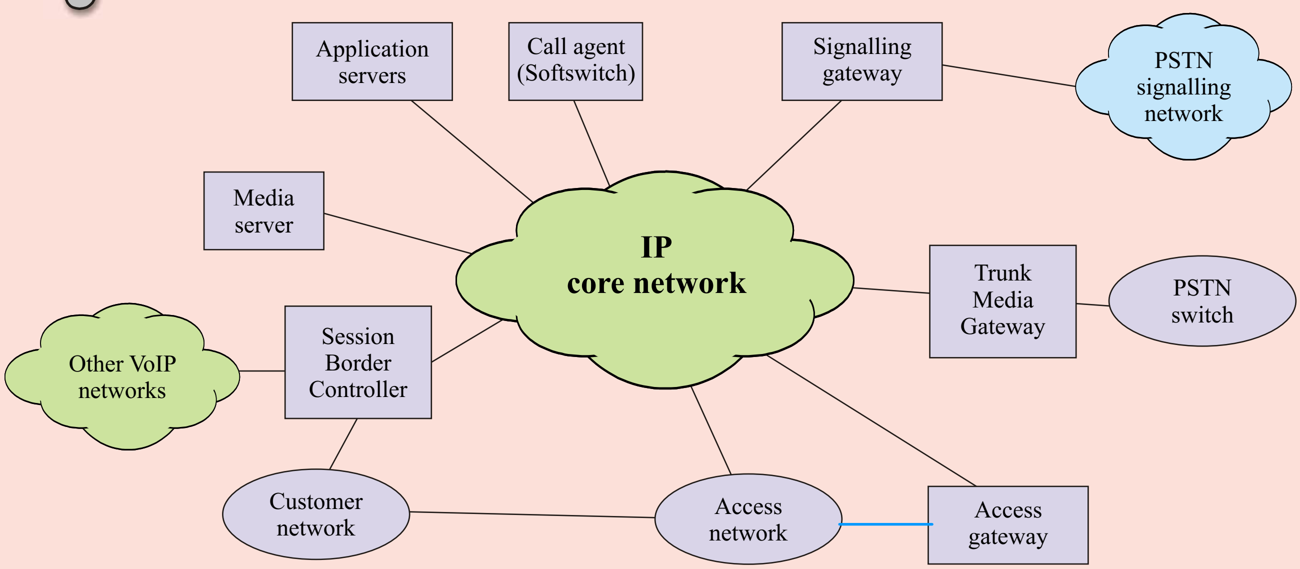
#### Services and Terminals

A crucial decision facing an operator considering to deploy a VoIP network is the service set that needs to be supported. This could range from a minimal set of low cost voice services offering possibly alongside broadband data services, through to full PSTN equivalence and advanced services for carriers wishing to replace their current infrastructure with a new converged network for all subscribers.

Another important part of the service design is the choice of end user terminals that are to be supported by the service offering. Possible choices include:

(contd....on page 4)

## VoIP Functional Architecture




**Access network (AN):** The access network provides connectivity between the customer premises equipment and the access gateways in the service provider's network. There are various access methods: TDM direct access, switched TDM, broadband access (cable, DSL), IP managed Internet service, etc.

**Access gateway (AG):** The AG is located in the service provider's network. It supports the line side interface to the core IP network for use by phones, devices, and PBXs. This element provides functions such as media conversion (circuit to Packet, Packet to circuit) and echo control.

**Trunk Media gateway (TMG):** The TMG supports a trunk side interface to the PSTN and/or IP routed flows in the packet network. It supports functions such as packetisation, echo control etc.

**Call agent (CA):** Call Agents are also known as media gateway controllers, Softswitches and call controllers. The CA is located in the service provider's network and provides call logic and call control functions, typically maintaining call state for every call in the network. Many CAs interact with application servers to supply services that are not directly hosted on CAs.

**Signalling gateway (SG):** The SG provides the signalling interface between the VoIP network and the PSTN signalling network. It terminates SS7 links and provides Message Transport Part (MTP) Level 1 and Level 2 functionality. Each SG communicates with its associated CA to support the end-to-end signalling for calls.

**IP core network:** The primary function of the IP core network is to provide routing and transport of IP packets. The IP core also has the added value of architecturally isolating the gateways, and their associated access networks, from the CA and associated service intelligence. In order to address the performance needs of each of the typical traffic streams associated with the VoIP architecture (bearer channels, signalling, and management traffic), the core network may support separate QoS mechanism.

**Media server (MS):** The MS is located in the service provider's network and uses a control protocol such as H.248 or SIP, under the control of the CA or application server, to provide announcements and tones, and collect user information.

**Application server (AS):** The AS is located in the service providers network, and provides the service logic and execution for one or more applications or services that are not directly hosted on the CA. Typically the CA routes calls to the appropriated AS for features the CA does not support.

**Session Border Controller (SBC):** It is deployed at the edge and core of a service provider's network to control signalling and media streams as they enter and exit the network. The “edge” is any IP-IP network border such as between a service provider and a customer or between a service provider and an enterprise network. The 'core' is any IP-IP network border such as those between two service providers. SBC provides functions such as security, denial of Service attacks, overload control, Network Address Translation and Firewall Traversal, Lawful Interception, Quality of Service (QoS) management, Protocol Translation, call accounting etc.

*Courtesy: IEEE Communications, July 2004*

- POTS "black phones"
- SIP phones (IP phones)
- PBXs and key systems
- PC soft-clients (including web-based applications)

Many of the issues discussed below are fundamentally affected by the decisions made in regard to services and terminals.

### Quality of Service (QoS)

One of the key requirements for the widespread deployment of VoIP would be the need to offer a toll quality service equivalent to the existing PSTN.

Quality of voice depends on the provision of bandwidth as per requirements and end-to-end delay.

Perceived Voice quality is very sensitive to key performance criteria in a packet network, in particular:

- Delay (latency) i.e. time taken to send a packet from a sending node to a receiving node.
- Jitter i.e. delay variation between packets arrived at a receiver compared to the packet spacing at the sender.
- Packet loss.

End-to-end delay in turn depends on speech coding techniques such as G.711 (64 kbps), G.726 (32 kbps), G.728 (16 kbps), G.729 (16 kbps) etc.; packetisation interval and jitter buffer delay. A one-way end-to-end delay of less than 150 ms is normally recommended for user satisfaction. Quality of voice is measured by Mean Opinion Score (MOS). An MOS of 4.0 or better (on scale of 1-5) is considered as 'toll quality'. Quality requirements are governed by Service Level Agreement (SLA) between the customer and the service provider.

IP, by its nature, provides a best-effort service and does not provide guarantees about the key criteria. Therefore, it is necessary to implement a suitable QoS solution in the majority of cases where simple over provisioning cannot guarantee success. There are a large number of technologies that can be chosen to provide QoS support such as Diffserv, Resource Reservation Protocol (RSVP) and Multi Protocol Label Switching (MPLS). However, the objective of such a solution is always to guarantee prioritization of voice media streams over best-effort data, and to ensure that the voice service is not compromised by unforeseen traffic patterns.

### Choice of Signalling Protocol(s)

Numerous different signalling protocols have been developed that are applicable to a VoIP solution. They include

- Media Gateway control protocols such as H.248 (Megaco), MGCP etc.
- Access services signalling protocols such as SIP, H.323 etc.
- Network service signalling protocols such as SIP, SIP-T, BICC etc.

The choice of which protocol to use in a service provider network is dependent upon both the service set being offered and the equipment available to provide these services. For example a network must support SIP in order to provide access to SIP phones.

### Security

The PSTN has been very resistant to security attacks and has not suffered from significant problems since the introduction of SS7 out-of-band signalling. A VoIP Next-Generation network is much more susceptible to security attacks and must address three key security issues.

#### *Denial of Service*

A denial of service attack prevents legitimate users of a network from accessing the features and services offered by that network. Denial of service attacks are extremely difficult in the PSTN but all too common in IP networks.

*Theft of Service*

Theft of service attacks are aimed at the service provider, where the attacker simply wants to use a service without paying for it.

Even in a VoIP access network using for example DSL, bandwidth is still a limited resource-especially the low packet loss and jitter required for good voice quality. Therefore, the network needs to be protected from subscribers misusing this high-priority bandwidth. One example would be if two SIP User Agents could set up a direct call between them, accessing the high priority bandwidth but bypassing the SIP Server(s) and hence not get billed.

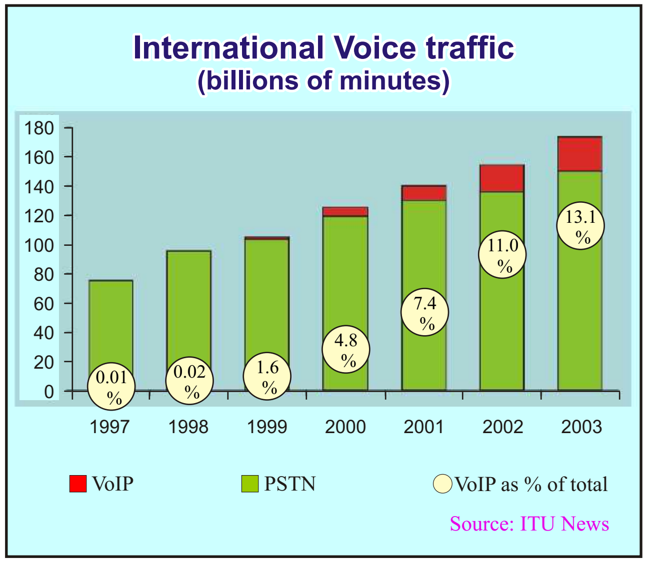
*Invasion of Privacy*

Subscribers to the PSTN expect that their calls are private, and that no third party can eavesdrop (with the exception of lawful interception). The PSTN achieves this privacy mainly by physical security mechanisms i.e. the wire from a subscriber's home is only connected to the local exchange or digital loop carrier and cannot easily be accessed.

This is not necessarily the case with VoIP networks, in particular, cable and wireless networks use a shared media which allow eavesdropping unless encryption is used. However, it is important to note that there is no "one size fits all" approach to security for VoIP. For example, networks that use an ATM based DSL access are fundamentally point-to-point networks and for these networks encryption is unnecessary, provided that the core network is suitably secured.

**Reliability / Availability**

The PSTN achieves five-nines reliability, equivalent to fewer than five minutes per year




downtime, and it handles millions of simultaneous calls. A VoIP network needs to achieve similar levels of reliability and scalability.

The required reliability and scalability can be achieved in a VoIP network by using redundant and load-sharing equipment and networks. The call agent, access gateway, trunk gateway, signalling gateway and media server need to be fault tolerant. The types of functionality often used to achieve fault tolerance include:

- Redundant hardware
- Redundant network connections
- Hot-swap capability
- No single point of failure
- Software and firmware that can be upgraded without loss of service

**Lawful interception**

Lawful Interception is an important regulatory requirement. VoIP networks typically contain separate call agents (Media Gateway Controller) and media gateways. The call agent is responsible for all call control and is the element that collects all the details of the calls required in an intercept. However, call content has to be collected elsewhere in the network.

The requirement to be able to tap the content of calls without the subscriber being able to detect any change implies that all calls, whether they remain within the carrier's IP network or access another network (e.g. PSTN) must be routed by the Call Agent via a device capable of duplicating the content and passing it to law enforcement.

### Emergency and Operator Services

The PSTN supports extensive Emergency and Operator Services. A Next-Generation VoIP Network needs to provide similar support leading to the following requirement:

- Support for legacy Emergency and Operator Services Interfaces, for example MF and SS7.
- Support for lifeline support where this is a regulatory requirement.
- Provision of location information so that a caller's physical location can be determined.

### Call Routing and Numbering Plans

The PSTN is able to route calls between telephones anywhere in the world. This is achieved by having a well-defined number plan both nationally and internationally. Routing tables can be built using this numbering plan to provide end-to-end connectivity.

A Next-Generation VoIP Network must provide the same capability, which requires the following:

- International and National numbering/ addressing plans, for example ENUM implementations
- Interconnection to the PSTN and E.164 numbers
- SIP endpoint addressing schemes
- Allocation of numbers/addresses
- Call routing between numbers/address

### DTMF and Other Tones and Telephony Events

When using VoIP there is an issue in transporting DTMF and other Tones and Telephony Events. These can flow transparently using a full rate codec such as G.711 but can't be transported using lower-bit codecs such as G.729.

There are several solutions used for transporting these tones and events but the most widespread are:

- Use RTP packets as specified by RFC 2833
- Transport the DTMF tones out of band using the signalling, e.g. SIP or H.248

### Fax, Modem and TTY support

Compared to voice traffic, fax, modem and TTY traffic is much more sensitive to packet loss but less sensitive to overall delay. In addition, lower-bit-rate codecs are optimized for voice traffic and cannot transport non-voice traffic.

ITU-T T.38 defines how fax can be sent in an IP network as pure data, independent of the voice traffic. However, it is a relatively recent standard and requires the use of a T.38 capable fax machine or T.38 gateway. The ITU-T has recently published equivalent standards for modems (V.150.0 and V150.1) and is currently working on developing equivalent standards for TTY traffic.

Alternatively fax, modem and TTY traffic can be supported successfully over a managed IP network by switching to a full rate codec (G.711). The media gateways need to detect a fax, modem or TTY call and switch to G.711. silence suppression and echo cancellation also need to be turned off.

Note that the detection and switch to G.711 needs to be performed in a timely manner, to allow the fax/modem to train at the highest possible data rate.

## Firewall and NAT traversal

For equipment that is resident at customer premises, such as IP phones and Subscriber Gateways, it is likely that there will be a firewall at the edge of the customer premises. In addition, Network Address Translation (NAT) may be used to convert internal IP addresses to external IP addresses.

Therefore, it is important that both the RTP media traffic and the signalling flows (SIP, H.248, MGCP) can negotiate both NAT and the firewall. For the firewall to be effective it needs to ensure that only authorized flows enter or leave the networks.

There are working groups within the IETF, including Midcom and NSIS, who are addressing the issue of communications with firewalls and network address translators.

## Billing and Reconciliation

The PSTN has extensive and accurate mechanisms for billing both subscribers and reconciliation between service providers.

A VoIP network must provide similar mechanisms to allow service providers to generate revenue. At least in the short-term it is likely that the existing billing mechanisms will remain in place both for inter-carrier reconciliation and subscriber billing, which requires generation of equivalent CDR records. Longer-term billing could move to be based on the bandwidth used, requiring alternative record keeping mechanisms such as those specified by IPDR.

## Network Interconnection

The PSTN is not a single network but a collection of networks operated by different service providers. At each network boundary a network interconnection is required. Network interconnection agreements are put in place to cover items such as interconnection points, signalling, timing, billing and tariffs, bearer transport, regulatory requirements, etc.

The next generation VoIP network will need to cover topics similar to existing interconnect agreements and also address additional items such as security, QoS, signalling protocols (SIP, SIP-T, BICC etc.) and deployment of Session Border Controllers.

### Internet Telephony in India

In India, Internet Service Providers (ISP) are allowed to provide Internet Telephony, restricted to connecting the following:

(a) PC to PC ; within or outside India

(b) PC in India to Telephone outside India

(c) IP based H.323/SIP Terminals connected directly to ISP nodes to similar Terminals; within or outside India.

Addressing scheme for Internet telephony shall only conform to IP addressing Scheme of Internet Assigned Number Authority (IANA) exclusive of National numbering Scheme/plan applicable to subscribers of Basic/cellular telephone Service.

It has been decided recently that licensed Access Service Providers can provide Internet Telephony, Internet services and Broadband services. If required, access service provider can use the network of NLD/ILD service licensee.

*Source: DoT website*

## Auto-configuration

One significant difference between a POTS (plain old telephone service) network and a Next-Generation VoIP network is that for some architectures, intelligent subscriber gateways or IP phones now reside on the customer premises.

These complex devices need more configuration than a POTS phone, so auto-configuration of subscriber gateways becomes important as the network scales up.

Some of these requirements can be addressed using DHCP (Dynamic Host Configuration Protocol), but others require some form of management interface using SNMP (Simple Network Management Protocol), LDAP (Lightweight Directory Access Protocol) etc.

Source : Extracts from MSF (msforum) white paper on 'VoIP Network Architecture' are used in this article.

## IMPORTANT ACTIVITIES OF TEC DURING SEPTEMBER TO NOVEMBER 2005

### Preparation of GRs/IRs

Following GRs/IRs and Technical documents were issued:

- Intelligent Peripheral for IN (IP-SRP)
- IR on Terminals for Connecting to PSTN
- Standard on Radio Device in unlicensed band 2.4 GHz
- Optical fibre Termination and Distribution Box
- MMSC for CDMA 2000 Network
- R-UIM for CDMA 2000 Network
- Session Border Controller
- Enterprise storage infrastructure

### Revised GRs/IRs

- High Speed Data Circuits Tester
- Standard on Primary Reference Clock Frequency
- High Speed Line Driver
- NSD/ ISD Payphones
- Cable, House Wiring PVC (Tinned Copper Conductor, PVC Insulated, Taped and PVC Sheathed)
- Digital Distribution Frame (DDF)

### Tests and Field trials

Tests/field trials have been carried out for:

- Fixed Line Prepaid service (FLPP) of BSNL project
- IN system of M/s Nortel at Coimbatore
- DWDM of M/s C-DoT
- IN system of East Zone IMPCS system at Kolkata

### Other Activities

**Manufacturer Forum conducted for**

- ADSL Test Set
- Ethernet Traffic Analyzer
- VRLA batteries based on GEL Technology
- 50W/100W/200W/300W Solid State Power Amplifier System operating in C-band
- 50W/250W Solid State Power Amplifier System operating in Ku-band
- Radio Products

Technical paper on 'Storage Data Centre for BSNL and MTNL' submitted to Telecom Commission

**Approvals issued by TEC during the period September 2005 to November 2005**

| | |
|---|---|
| Interface Approvals | 42 |
| Service Test Certificate | 46 |
| **Total** | **88** |

**Approvals issued by TEC upto 30.11.2005**

| | |
|---|---|
| Type Approvals | 7018 |
| Interface Approvals | 4221 |
| Service Test Certificate | 1758 |
| **Grand Total** | **12997** |



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