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# Next-Generation VoIP Network Architecture


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# ABSTRACT

This white paper is part of series of white papers that will be published by the MSF in 2003. The series of white papers explore the issues motivating the MSF 2003 work plan. The white papers also discuss the solution space within which the MSF will work and identify a proposed work program for 2003. The end goal of the 2003 work program will be a second Global MSF Interoperability (GMI2004) event.

This White paper identifies and characterizes the primary issues that must be addressed in defining a large scale VoIP network that is capable of supporting, but is not necessarily limited to, full PSTN equivalence. It also provides the foundation for the other white papers in the series.

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# Executive Summary

The term "Voice over IP" (VoIP) describes the transport of voice over IP based networks, it is a generic term that covers deployments ranging in complexity from hobbyists using the internet to get free phone calls on a peer to peer basis, to full scale PSTN replacement networks. In carrier networks VoIP has been mainly deployed in enterprise networks or as a trunking technology to reduce transport costs in voice backbone networks. However with many network operators facing eventual equipment obsolescence in their existing narrowband PSTN networks and with the drive to increase revenue by offering new and innovative multimedia services, the MSF expects that end-to-end VoIP solutions will be required to replace the PSTN in the medium term.

In order to deploy a VoIP network that is capable of providing a PSTN scale solution the following issues must be addressed.

- What services need to be offered, for example full PSTN equivalence, a more restricted "cheap second line" service, or a simple user-to-user voice service. This white paper will focus on a service set that provides full PSTN equivalence – what we are calling "Telephony over IP".
- The types of end user terminals supported – POTS phones, PC clients, IP Phones or PBXs.
- Quality of Service requirement for voice to ensure that the agreed quality is provided.
- The security risks must be clearly identified and appropriate techniques employed to ensure that the call agents, in particular, are protected from attack
- How much bandwidth is available in the last mile network, which will affect the choice of voice codec, packetization period, and where to use compression to best meet the service goals.
- The signaling protocol used must support the service set required.
- Lawful interception requirements in many countries could prevent a public carrier from allowing direct connection between IP phones. All calls may need to be routed via an access gateway that hides any intercepts in place.

The objective of this White paper is to identify and characterize the primary issues that must be addressed to define a large scale VoIP network that is capable of supporting (but not necessarily limited to) full PSTN equivalence. Other MSF White Papers will explore these issues in greater detail, and help define a technical program to develop additional Implementation Agreements to support full multi-vendor interoperability. This process will culminate in a second multi vendor interoperability event (Global MSF Interoperability 2004).

# Introduction

This white paper outlines the requirements, benefits and issues involved in migrating to a Next-generation Network using VoIP. It presents a high-level network architecture that acts as a starting point from which PSTN equivalent VoIP networks can be built. It then identifies work items that will be followed by the MSF technical committees as they seek to refine the solution.

The goal of the MSF is to promote multi-vendor interoperability as part of a drive to accelerate the deployment of next generation networks. To date the MSF has defined a number of detailed Implementation Agreements and detailed Test Plans for the signaling protocols between network components. Moving forward the MSF will seek to develop additional Implementation Agreements and Test Plans addressing some of the other technical issues such as QoS and Security to assist vendors and operators in deploying interoperable solutions. To this end the MSF will look to adopt pragmatic solutions in order to maximize the chances for early interoperability.

The MSF welcomes feedback and comment and would encourage interested parties to get involved in this work program. Information about the MSF and membership options can be found on the MSF website http://www.msforum.org/

# 1 Overview of VoIP

At its simplest, Voice over IP 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 phone calls over the public internet. 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 together, e.g. a low cost way of calling the USA from the UK. 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.

This white paper addresses the central question of how would a carrier deploy a voice over IP service that offered end to end VoIP whilst offering PSTN equivalence. It considers the issues faced by such a carrier and how the end to end VoIP service fits into the legacy PSTN infrastructure that will undoubtedly remain for a good many years.

# 2 Key Benefits and Requirements for VoIP

For service providers examining the business case for VoIP, the ubiquity of IP as a networking technology at the customer premises opens the possibility of deploying a vast range of innovative converged voice and data services that simply cannot be cost effectively supported over today's PSTN infrastructure.

- IP-based internet applications, such as email and unified messaging, may be seamlessly integrated with voice applications
- IP centrex services allow network operators to provide companies with cost effective replacements for their ageing PBX infrastructure
- VoIP services can be expanded to support multimedia applications, opening up the possibility of cost effective video conferencing, video streaming, gaming or other multi-media applications.

What is more the traditionally open interfaces and enterprise culture surrounding IP networking results in reductions in cost, and gains in productivity and time to market for service providers.

- The flexibility of next generation platforms allows for the rapid development of new services and development cycles are typically shorter than for ATM or TDM-based equipment.
- VoIP products based on the MSF architecture, unlike legacy TDM switches, often support open service creation environments that allow third party developers to invent and deliver differentiated services.
- Third parties may develop applications and services for end users on open architecture CPE devices such as PCs. By co-operating with such third parties network operators stand to gain increased revenues from the explosion of innovative services that this advance is likely to trigger.

In addition the consolidation of voice and data in one network can significantly reduces cost.

- VoIP leverages data network capacity removing the requirement to operate separate voice and data networks.
- IP equipment is typically faster and cheaper than ATM or TDM-based equipment – a gap that is increasing rapidly every few months.
- Re-routing of IP networks (e.g. with MPLS) is much cheaper than, say, SDH protection switching.

Whatever the justifications, most service providers recognize that VoIP is the direction of the future – however when looking at a future PSTN scale solution service providers must ensure that the following key requirements are met to provide equivalence with the PSTN:

- Security
- Quality of Service
- Reliability
- Migration path
- OSS support
- Billing
- Network Interconnection

These issues are by no means simple and in many cases have delayed roll out of VoIP services. This white paper will look in more detail at these problems and consider at a high level how they might be addressed.

# 3 VoIP Next-Generation Network Architecture

VoIP can be deployed in many different network segments. To date, it has been mostly deployed in the backbone and enterprise networks. Deploying VoIP as an end-to-end Next-Generation Network solution introduces additional constraints and issues discussed in section 5.

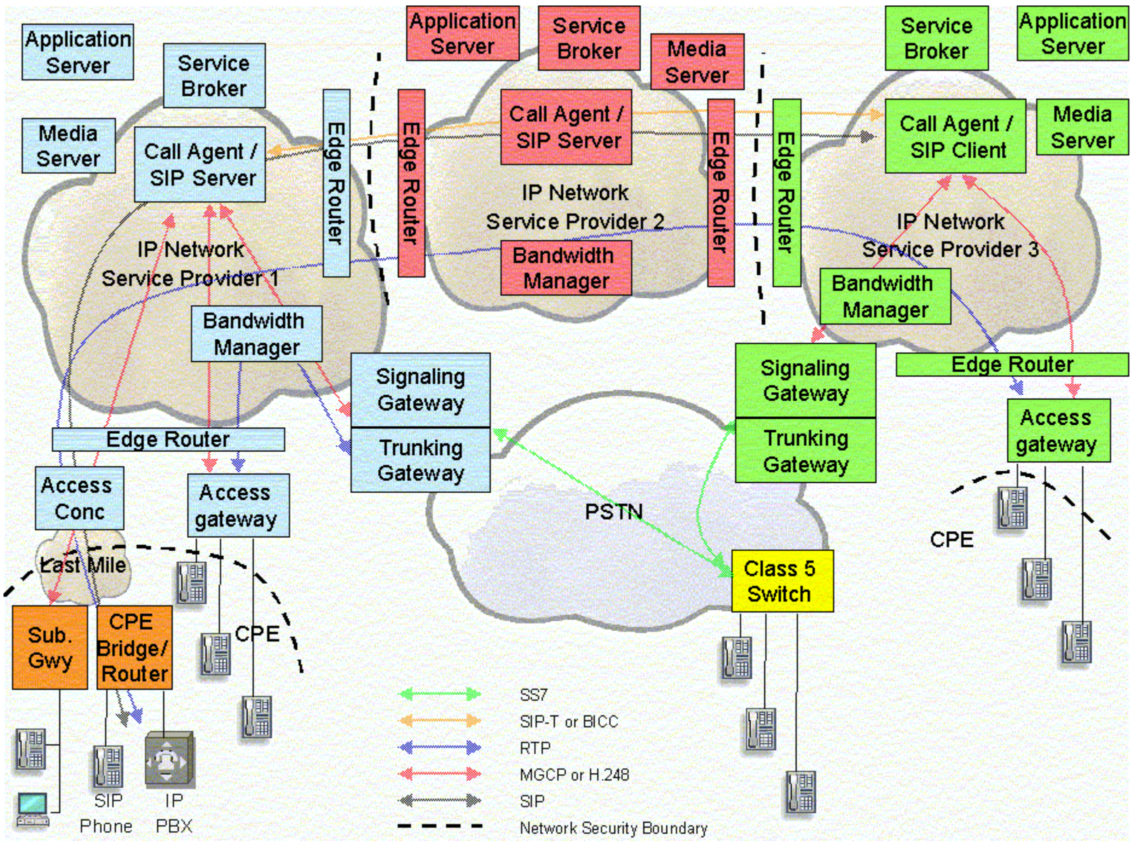


**Figure 1: Next Generation VoIP Network**

Figure 1 shows an example VoIP Next Generation network with 3 service provider networks.

- Service Provider 1 is offering local access acting as a LEC. This Service Provider supports IP phones and IP PBX systems using SIP and POTS phones via either an Access Gateway (Next-Gen DLC) or a Subscriber Gateway (using either H.248 or MGCP).
- Service Provider 2 is acting as an inter-exchange carrier (IXC) and supports SIP and SIP-T or BICC signaling through its network.
- Service Provider 3 is offering local access acting as a LEC, but only supports POTS phones using an Access Gateway. SIP signaling is supported but is terminated by the SIP Server rather than using a SIP Phone or other CPE device.

# 4 Network Components

This section describes the function of the network components listed in Figure 1. Depending upon the particular network architecture some of these network components may be combined into a single solution, for example a combined signaling and trunking gateway.

## 4.1 Call Agent/SIP Server/SIP Client

The Call Agent/SIP Server/SIP Client 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 call agents include service logic for supplementary services, e.g. Caller ID, Call Waiting, and also interact with application servers to supply services that are not directly hosted on call agent. The Call Agent will participate in signaling and device control flows originating, terminating or forwarding messages. There are numerous relevant protocols depending upon the network architecture including SIP, SIP-T, H.323, BICC, H.248, MGCP/NCS, SS7, AIN, ISDN, etc. Call Agents also produce details of each call to support billing and reconciliation.

A SIP Server provides equivalent function to a Call Agent in a SIP signaling network, its primary roles are to route and forward SIP requests, enforce policy (for example call admission control) and maintain call details records. For example the SIP Server in Service Provider 1's network will route and forward SIP requests from SIP Phones belonging to customers.

A SIP Client provides similar function to a SIP Server, but originates or terminates SIP signaling rather than forwarding it to a SIP Phone or other CPE device. For example a SIP call is shown in figure 1 between a SIP Phone in Service Provider 1's network and the SIP server in Service Provider 3's network. The Call Agent/SIP Server terminates the SIP signaling and converts it to H.248 or MGCP to set up a call to the correct subscriber.

Call Agents are also known as Media Gateway Controllers, Softswitches and Call Controllers. All these terms convey a slightly different emphasis but maintaining call state is the common function.

## 4.2 Service Broker

The service broker is located on the edge of the service providers service network and provides the service distribution, coordination, and control between application servers, media servers, call agents, and services that may exist on alternate technologies (i.e. Parlay Gateways and SCP s). The service broker allows a consistent repeatable approach for controlling applications in conjunction with their service data and media resources to enable services to allow services to be reused with other services to create new value added services.

## 4.3 Application Server

The Application Server is located in the service provider's network and provides the service logic and execution for one or more applications or services that are not directly hosted on the Call Agent. For example, it may provide voice mail or conference calling facilities. Typically the Call Agent will route calls to the appropriate application server when a service is invoked that the Call Agent cannot itself support.

## 4.4 Media Server

This Media Server is located in the service provider's network. It is also referred to as an announcement server. For voice services, it uses a control protocol, such as H.248 (Megaco) or MGCP, under the control of the call agent or application server. Some of the functions the Media Server can provide are

- playing announcements
- mixing – providing support for 3-way calling etc
- codec transcoding and voice activity detection
- tone detection and generation
- interactive voice response (IVR) processing
- fax processing.

## 4.5 Signaling Gateway

The Signaling Gateway is located in the service provider's network and acts as a gateway between the call agent signaling and the SS7-based PSTN. It can also be used as a signaling gateway between different packet-based carrier domains. It may provide signaling translation, for example between SIP and SS7 or simply signaling transport conversion e.g. SS7 over IP to SS7 over TDM.

## 4.6 Trunking Gateway

The Trunking Gateway is located in the service provider's network and as a gateway between the carrier IP network and the TDM (Time Division Multiplexing)-based PSTN. It provides transcoding from the packet-based voice, VoIP onto a TDM network. Typically, it is under the control of the Call Agent / Media Gateway Controller through a device control protocol such as H.248 (Megaco) or MGCP.

## 4.7 Access Gateway

The Access Gateway is located in the service provider's network. It provides support for POTS phones and typically, it is under the control of the Call Agent / Media Gateway Controller through a device control protocol such as H.248 (Megaco) or MGCP.

## 4.8 Access Concentrator

The Access Concentrator is located in the service provider's network and terminates the service provider end of the WAN links used over the "last mile". For example, in a DSL network, this is a DSLAM; in a cable network, a CMTS.

The Access Concentrator may also include the Access Gateway function, for example a Next-Generation DLC that combines DSLAM capability with direct POTS termination.

## 4.9 Bandwidth Manager

The Bandwidth Manager is located in the service provider's network and is responsible for providing the required QoS from the network. It is responsible for the setting up and tearing down of bandwidth within the network and for controlling the access of individual calls to this bandwidth. It is responsible for installing the appropriate policy in edge routers to police the media flows on a per call basis.

## 4.10 Edge Router

The Edge Router is located in the service provider's network and routes IP traffic onto the carrier backbone network. Typically the edge router will provide many other functions and can be combined with the Access Concentrator.

## 4.11 Subscriber Gateway

The Subscriber Gateway is located at the customer premises and terminates the WAN (Wide Area Network) link (DSL, T1, fixed wireless, cable etc) at the customer premises and typically provides both voice ports and data connectivity. Usually, it uses a device control protocol, such as H.248 (Megaco) or MGCP/NCS, under the control of the Call Agent. It provides similar function to the Access Gateway but typically supports many fewer voice ports.

Subscriber Gateways are also known as IADs, Residential Gateways, or MTAs (in a cable network).

## 4.12 Bridge/Router

The Bridge/Router is located at the customer premises and terminates the WAN (Wide Area Network) link (DSL, T1, fixed wireless, cable etc) at the customer premises. The difference between this and the Subscriber Gateway is a bridge/router does not provide any native voice support, although voice services for example SIP phones, can be bridged/routed via this device.

## 4.13 IP Phone/PBX

IP Phones and PBX systems are located at customer premises and provide voice services. They interact with the Call Agent/SIP Server using a signaling protocol such as SIP, H.323 or a device control protocol such as H.248 (Megaco) or MGCP.

# 5 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. These include:

- Service set to be offered, and the types of end user terminal supported.
- Choice of signaling protocol(s).
- Security.
- Quality of Service (QoS).
- Reliability / availability.
- Regulatory Issues
    - Lawful Interception
    - Emergency and Operator Services
- Call routing and Number Plans.
- DTMF and Other Tones and Telephony Events
- Firewall and NAT traversal.
- Billing and Reconciliation.
- Network Interconnection.
- Migration Path.
- OSS support.
- Bandwidth Utilization.
- Fax, Modem, and TTY support.
- Auto-configuration.

## 5.1 Service set

A crucial decision facing an operator looking to deploy a VoIP network is the service set that needs to be supported. This could range from a minimal set of services for a "cheap teen line" 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:

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

This whitepaper and the MSF work program is aimed at providing PSTN equivalence service, but does not explicitly limit the choice of terminals. This decision does effectively define much of the problem space that the network designer will have to work with. This can be seen from the fact that many of the issues discussed below are fundamentally affected by the decisions made in regard to services and terminals.

## 5.2 Choice of Signaling Protocol(s

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

- Device control protocols such as H.248 (Megaco), MGCP, NCS, etc
- Access services signaling protocols such as SIP, H.323, etc
- Network service signaling protocols such as SIP, SIP-T, BICC, CMSS, 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.

## 5.3 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 signaling. A VoIP Next-Generation network is much more susceptible to security attacks and must address three key security issues.

- Denial of service
- Theft of service
- Invasion of privacy

Security is seen as a priority for MSF, and will be addressed in our 2003 work program. See section 6 for more details.

### 5.3.1 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. There have been several successful attacks on web servers on the Internet, even including the high security government sites.

In a complex network, there are many possible denial of service attacks. Some examples include sending false signaling messages so that a call agent is fooled into believing that a party has gone on-hook, bombarding a device with pings or other packets so frequently that it has no spare processing power to process legitimate requests and hacking a Subscriber Gateway to send ftp or other data traffic as high priority voice traffic.

### 5.3.2 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. The most common form in the current PSTN is called subscriber fraud, where a subscriber sets up an account with a service provider using false billing information, for example a stolen credit card. Other forms of theft are more technical, often utilizing black boxes or similar to fool the network into providing free service. It is interesting to note that fraudulent long-distance calls were more common when the network used in-band DTMF signaling which could be mimicked using a blue box.

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.

### 5.3.3 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.

## 5.4 Quality of Service

One of the key requirements for the widespread deployment of VoIP is the ability to offer a toll quality service equivalent to the existing PSTN. Indeed some carriers are even looking for Next-Generation Networks as a means for delivering much higher voice quality as a service.

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

- Delay
- Jitter
- Packet loss

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, RSVP,MPLS and even ATM. 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.

## 5.5 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, signaling 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.

## 5.6 Lawful Interception

Historically, lawful interception (wiretapping) of telephone conversations has been a relatively well-defined and straightforward process. Typically, a law enforcement agency applied to a court for an order to tap a particular phone number. Once the agency had the order, it served that order on the provider of the telephone service for the number to be tapped. The service provider then put a tap on the circuit, extracted all the necessary information and passed it to the law enforcement agency. The introduction of VoIP complicates this process considerably.

The law varies according to location (in the United States, the relevant legislation is the Communications Assistance for Law Enforcement Act - CALEA). The following requirements are typical for any network including VoIP networks and the PSTN.

- No wiretap is permitted without a court order.
- Wiretaps apply to phone numbers, not particular suspects.
- Wiretaps fall into two categories.
    - Call detail – a tap in which the details of the calls made and received by a subscriber are passed to the law enforcement agency. (Referred to as pen register and trap and trace in the U.S.).
    - Call content – a tap in which the actual contents of a call are passed to the law enforcement agency.
- The suspect must not detect the tap, so the tap must occur within the network and not at the subscriber gateway. Also, the tap may not be detectable by any change in timing, feature availability or operation.
- A suspect may be tapped by more than one agency. The taps are separate, and the various agencies are not aware of each other's taps. The taps do not have to be of the same category.
- It is the responsibility of the telecommunications carrier that originates or terminates calls to provide lawful interception.

As described in section 3, VoIP networks typically contain separate call agents 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 a call detail tap. However the call agent does not see the call content, so call content must 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 leads to the conclusion 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.

## 5.7 Emergency and Operator Services

The PSTN supports extensive Emergency and Operator Services. Subscribers can dial 911 or the local equivalent and reach Emergency Services under almost any conditions. A Next-Generation VoIP Network needs to provide similar support leading to the following requirements:

- 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.

## 5.8 Call Routing and Number Plans

The PSTN is able to route calls between telephones anywhere in the world, for example a user can call Australia from Canada. 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 and number portability issues
- Call routing between numbers/addresses

## 5.9 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 code 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 signaling protocol, e.g. SIP or H.248.

## 5.10 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 signaling 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.

## 5.11 Billing and Reconciliation

The PSTN has extensive and accurate mechanisms for billing both subscribers and reconciliation between service providers. Currently most billing mechanisms are based on usage, e.g. per minute billing, although some services are charged on a flat-rate basis, e.g. local calls in the US.

Service Providers generate Call Detail Records (CDRs) for traffic entering or leaving their networks and generate bills based on these.

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.

## 5.12 Network Interconnection

The PSTN is not a single network but a collection of networks operated by thousands of 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, signaling, timing, billing and tariffs, bearer transport, regulatory requirements, etc. In addition these normally require approval from the relevant regulator.

Constraints of scalability and established business models mean the next generation VoIP network will, like the PSTN, be a collection of networks and network interconnection agreements will still be required. For example in figure 1, Service Provider 1 and 2 will need an interconnect agreement as will Service Providers 2 and 3. This will need to cover similar topics to existing interconnect agreements but in addition address items such as security, QoS, signaling protocols (SIP, SIP-T, BICC). These additions may require regulatory approval as well.

## 5.13 Migration Path

While the eventual goal is an end-to-end Next-Generation Network, it will be decades before legacy networks disappear. On the access side, this means that ongoing support for POTS telephone lines and DLCs may be a requirement; in the backbone network, interconnection with SS7 signaling and TDM trunks, 911 and operator services, databases for 1-800 and local number portability and CALEA, are all essential. In addition migration will happen piecemeal in different carrier networks and individual service providers may support both next-generation and legacy networks in parallel.

Therefore it is crucial that Next-Generation Network equipment provides support for legacy networks and that interworking between the networks is reliable and flawless. Service provides must also carefully plan migration strategies that include both introducing new services and support for legacy interfaces.

## 5.14 OSS Support

The existing PSTN has very extensive Operations Support Systems providing such functions as

- Flow-through provisioning
- Fault isolation
- Loop testing
- Alarms
- Performance monitoring
- Policy definition and enforcement

A Next-Generation VoIP network will need to offer similar levels of OSS support. In addition given the huge investment in existing OSS systems any new equipment will need to be integrated with these which may require support for protocols such as CORBA, SNMP, TL1, etc

VoIP networks also introduce new requirements such as the ability to dynamically measure end-to-end voice quality.

## 5.15 Bandwidth Utilization

In a VoIP network digitized voice is transported using real-time protocol (RTP). A typical voice sample is less than 100 bytes, but the combined headers are at least 40 bytes. For lower-bandwidth WAN links such as DSL or Cable, the header overhead is significant and reduces the number of voice channels or data bandwidth available. Given that one of the advantages of using VoIP is that it should be possible to use lower bit codecs to save bandwidth, a mechanism for reducing the overhead is required.

The main approach to reducing the overhead is to implement compression for RTP, UDP and IP headers. However this requires a point-to-point link and the endpoints to maintain state for each compressed RTP flow.

## 5.16 Fax, Modem and TTY support.

The PSTN reliably supports fax, modem and TTY calls. Calls connect on almost every attempt and rarely fail. A VoIP network must provide a similarly reliable service. However, fax, modem and TTY traffic imposes some additional constraints beyond voice traffic.

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.

## 5.17 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, but others require some form of management interface using UPnP, SNMP or LDAP.

Considerable work has been done in the DSL Forum to address auto-configuration of DSL equipment, but to date the issue of auto-configuration in VoIP networks has not been addressed.

# 6 MSF VoIP Work Plan

The MSF is committed to an aggressive technical program to specify and prove a solution for a full scale PSTN replacement network over next generation IP infrastructure. This technical program will attack head on the major issues that have so far prevented the next generation network (NGN) from being fit for purpose, specifically it aims to provide for a network that is capable of offering Quality of Service (QoS) and security in a way that will scale to the many billions of busy hour calls that a typical PSTN must handle.
The MSF work program will follow the established and proven MSF approach to problem solving by:

- Proposing a coherent and pragmatic network vision.
- Identifying existing protocols that can deliver the vision and enhancing and profiling them so as to make them fit for purpose whilst removing barriers to interoperability
- Developing a program of early interoperability testing with carrier grade equipment supported by detailed and relevant test plans.

From a technical perspective the MSF has currently identified the following key technical areas:

- Scaleable IP QoS for Voice (and multimedia) over IP
    - This work area will define a solution for providing QoS over an IP network in a way that will scale to the many billions of BHCA that a PSTN network must support. As part of this work area the interaction between SIP services and QoS mechanisms such as MPLS-TE will be addressed and the functionality required in IP edge routers will be defined.
- Security in an end to end VoIP network
    - The nature of an end-to-end VoIP solution means that it is vulnerable to a variety of attacks, ranging from theft of service to denial of service. This work area will define a security framework for an MSF next generation network that will provide protection against malicious attacks.
- Service and Feature Interactions
    - This work area will further define the MSF service layer with a view to identifying and proving new and innovative network services and verifying that these services can be supported over the scaleable next generation network being defined by the MSF.
- Call routing for end to end VoIP
    - This work area will look at call routing mechanisms and the addressing issues that must be considered when working with a pure VoIP environment and a mixed VoIP, Legacy PSTN environment.
- Management
    - This work area will address management issues related to an MSF next generation network.

The MSF will publish a number of white papers that address these issues and these white papers will identify the basic solution space from which the detailed solution design will develop. The detailed design will motivate the overall architecture and act as a starting point for protocol profiling and interoperability testing.

# 7 Conclusion

IP is ubiquitous and cost-effective. By moving to a VoIP network a carrier can:

- Deploy new converged voice and data services
- Remove the need to manager separate voice and data networks
- Utilize cheaper IP-based backbone equipment to carry voice
- Reap the benefits of a standards-based and highly flexible network architecture, giving a competitive market between equipment vendors and encompassing a wide range of equipment for different market niches.

While there are a number of VoIP solutions available today, most of these have limitations of one kind or another. In some cases the solutions are built on early versions of standards and provide restricted interoperability with other vendors. In some cases the solutions do not provide the scalability, robustness, security or features required for PSTN equivalency. The MSF is committed to providing a next generation network that provides both full multi-vendor interoperability, and support for a full featured, secure PSTN service.

The MSF Release 1 Architecture and GMI 2002 event was focused on specifying VoIP for a Gateway scenario. The next step is to develop a coherent solution for scaleable next generation networks that support end-to-end VoIP. This will build on the proven methodology the MSF used for Release 1. It will identify open interfaces and define Implementation Agreements for these interfaces. The MSF will then produce test plans and conduct interoperability testing to accelerate the deployment of next generation networks. It is the aim of the MSF to shorten the timescales in which end-to-end VoIP solutions become available, and to accelerate the transition of carrier TDM voice networks to VoIP networks.

# 8 Acronyms

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| BICC | Bearer Independent Call Control |
| BRI | Basic Rate Interface |
| CA | Call Agent |
| CDR | Call Detail Records |
| CORBA | Common Object Request Broker Architecture |
| DHCP | Dynamic Host Control Protocol |
| DSL | Digital Subscriber Line |
| IAD | Integrated Access Device |
| IP | Internet Protocol |
| ISDN | Integrated Services Digital Network |
| LDAP | Lightweight Directory Access Protocol |
| MEGACO | Media Gateway Control Protocol |
| MGC | Media Gateway Controller |
| MGCP | Media Gateway Control Protocol |
| MTA | Multimedia Terminal Adaptor |
| NAT | Network Address Translation |
| NCS | Network-based Call Signaling |
| OSS | Operations Support Systems |
| POTS | Plain Old Telephony System |
| PRI | Primary Rate Interface |
| PSTN | Public Switched Telephone Network |
| QoS | Quality of Service |
| SIP | Session Initiation Protocol |
| SIP-T | Session Initiation Protocol for Telephones |
| SNMP | Simple Network Management Protocol |
| SS7 | Signaling System Number 7 |
| TDM | Time Division Multiplexing |
| TGW | Trunking Gateway |
| TL1 | Transaction Language 1 |
| TTY | Teleprinter/Teletype/Teletypewriter |
| UPnP | Universal Plug and Play |
| VoIP | Voice over IP |