Roger Ward

Results of the MSF Global Interoperability Programme (GMI 2004)

The MultiService Forum (MSF) is well established within the global industry, with a successful track record spanning over six years since its formation in 1998. With BT and other major carriers as founding members, it is perhaps not surprising that the work of the MSF has developed over the years to fully reflect the needs of major carriers. Enjoying now a large membership comprising many leading network operators and global suppliers of telecommunication network equipment, the MSF provides a unique environment for constructive collaboration. Its mission—to accelerate the deployment of open communications systems that realise the economic benefit resulting from the flexible support of a full range of network services using multiple infrastructure technologies, more relevant today than ever before.

This article presents the results of the Global MSF Interoperability Programme (GMI2004) – its scope, organisation and achievements. As earlier articles have discussed more fully[1, 2] the MSF provides a unique collaboration framework for forward-looking organisations that see the competitive advantage that can be achieved by collaborating on issues of common concern. With a meaningful agenda formulated from over six years experience, the MSF is at the forefront of leading carriers' plans to implement the networks of the 21st century today. As one of the most comprehensive and advanced network-level interoperability events ever undertaken, GMI2004 was able to bridge North America, Europe, and Asia pushing voice over IP (VoIP) and multi-protocol label switching (MPLS) interoperability testing into new territory.

By creating an aggressive multi-vendor test environment that included emerging and existing technologies, GMI2004 provided realistic test scenarios that included multiple global carriers and many increasingly important multiservice and transport protocols, such as session initiation protocol (SIP), Internet Protocol Version 6 (IPv6), and H.248. From architecture, through protocol profiling and global MSF interoperability testing, the MSF provides a cost-effective framework to develop implementation agreements (IAs) which are of benefit to carriers and suppliers alike. The results of GMI2004 described here are tangible proof of the success of the MSF's approach and underpin the MSF ongoing 2005/6 work programme, ongoing work that is dedicated to the relationship of fixed and mobile networks and the delivery of profitable services and applications.

Introduction

The MSF was formed in 1998 to develop and promote an open architecture and enhanced interoperability for the telecommunications industry. The forum was created in response to rapidly changing industry conditions: revenues from traditional sources, like voice, were declining, yet operating expenses continued to climb. Carriers began looking for ways to deliver new profitable services while reducing costs. Multivendor interoperability was a critical component of both goals. At the time, however, carriers often purchased equipment from a single vendor to ensure interoperability. The MSF set out to change the status quo.

The cost of demonstrating a truly interoperable, next-generation global IP network is staggering. For example, GMI 2004 spent more than $2 million on 16 800 man-hours of testing, and that figure does not include lab resources and equipment. Even if a vendor or system supplier were willing to invest that sort of money, it would still have to hire staff and purchase the

The Author: Roger Ward is with BT. This article is based on the 'The Global MSF Interoperability (GMI) 2004 White Paper' which is reproduced by kind permission of The MultiService Forum.

necessary equipment, a daunting financial proposal.

There may be one or two companies that actually have the resources required to mount a full-scale, global interoperability evaluation. Thanks to the MSF, however, there's no need to incur those sorts of expenses. GMI events are made possible, at least in part, by the generosity of the participants, who donate equipment, on-staff expertise, lab resources, and even space in their network operation centres (NOCs).

What do carriers and equipment vendors get in return? Those that actively participate in GMI events learn how multivendor next-generation products and networks will interoperate in the real world. That information translates into several financial benefits:

- reduced time to market for deployment of an interoperable solution,
- decreased costs and resources to resolve interoperability issues,
- improved protocol documentation through clarifications in the MSF IAs and standards process, and
- thoroughly evaluated architectural framework for cooperatively designing end-to-end networking solutions.

The GMI 2004 Programme

GMI 2004 did far more than merely build a next-generation multiservice network. It established that a variety of vendors can already make good on the promise of next-generation services. It proved that carriers and providers can deliver VoIP and other enhanced services globally. It showed that products used in the event are ready to take their place on next-generation telephony networks.

In addition, GMI 2004 demonstrated:

- a scalable next-generation network that can generate revenue from multiple services,
- value-added services provided by Parlay gateways and application servers,
- a global infrastructure that can deploy voice, video, and data while supporting both current and emerging services and applications,
- enhanced quality of service (QoS), and
- global interoperability of the MultiService Forum (MSF) architecture as detailed in Release 2.

GMI 2004 required more than 10 months to plan and execute. Its multivendor multiservice network spanned four countries and three continents. The demonstration employed more than 50 test plans to exercise the 25 new MSF implementation agreements (IAs) that had been introduced since the original GMI 2002. Internetwork connections ranged from tried-and-true ATM to leading-edge IP on Internet2.

The result was a stunning success that met or exceeded all objectives. Over the course of 12 days, four global carriers and 28 major equipment and network management vendors demonstrated interoperability under real-world conditions, along with the 'five 9s' uptime crucial to carrier-class deployments.

Benefits of MSF Work and Architecture

The benefits of the MSF's technical work are clear: cost savings and future-proofing. The MSF architecture and solution framework combine legacy and next-generation services in a single unified network. Further, since all MSF participants implement the same baseline features and functions, members can eliminate the guesswork that technology development typically involves.

Service providers and system suppliers also have the opportunity to learn from the experience of some of the world's leading scientists and engineers. Network equipment vendors profit by pooling resources and working together on implementations of standard protocols and architectures.

GMI 2004: 28 partipants

Acme Packet
Agilent Technologies
Alcatel
British Telecommunications plc
Cisco Systems
Convedia Corporation
Empirix
Electronics and Telecommunications Research Institute
Ericsson
FEELingK
Fujitsu Network Communications, Inc.
Hitachi
IP Unity
KT
Leapstone
Marconi plc
MetaSwitch
National Communications System
Navtel Communications Inc.
Nortel Networks Ltd.
NTT
Operax
Qwest
Siemens
Softfront
Sonus Networks
Spirent Communications
Teledata Networks

Participants in GMI 2004

The MSF counts some of the leading global carriers among its members, along with the telecom industry's top vendors and most exciting startups (see panels below).

The GMI Network

GMI 2004 took place from 4 October to 16 October 2004. Its global test bed was based at four geographically dispersed sites:

- Adastral Park, the BT Advanced Research and Technology Centre, Martlesham, Ipswich, UK;
- NTT Musashino Research and Development Center, Tokyo, Japan;
- KT Research and Development Labs, Daejon, South Korea; and
- Qwest Communications Lab, Dublin, Ohio, USA.

MSF membership – 2005

Acme Packet
Alcatel
British Telecommunications plc
Cable & Wireless
Cisco Systems
Convedia Corporation
ECI Telecom Ltd.
Empirix
Electronics and Telecommunications Research Institute
Ericsson
Fujitsu Network Communications, Inc.
Hitachi
Huawei Technologies
IP Unity
Italtel
KT
Leapstone
LG
Marconi plc
MetaSwitch
Mitsubishi Electric
National Communications System
Navtel Communications Inc.
NEC Corporation
Netrake
Newport Networks Ltd
Nortel Networks
NTT
Operax
Samsung Electronics
SBC
Siemens
Spirent Communications
Tdsoft
Tekelec
Teledata Networks
Veraz Networks
Verizon Communications
Vodafone Group plc
Xener Systems
ZTE Corporation

BT Global Services interconnected three sites, with dedicated ATM links terminated in London, Philadelphia, and Seoul. In the UK, BT also provided the fibres that connected the London point of presence (POP) to Adastral Park. In the US Qwest supplied the connecting fibre link in Philadelphia. Finally, KT furnished fibre connectivity from the Seoul POP to Daejon, Korea. This link provided an 8 Mbit/s constant bit rate (CBR) to three of the four host sites.

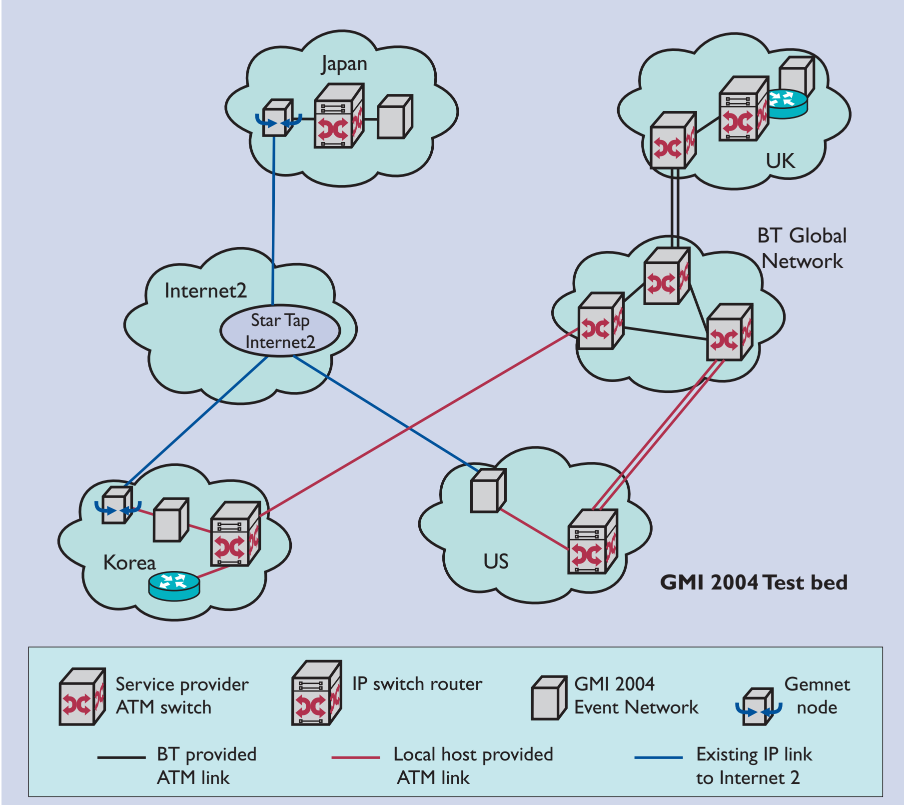
Internet2® also supplied connectivity to the demonstration network. Internet2 is a consortium led by 207 universities working in partnership with industry and government to develop and deploy advanced network applications and technologies. The Internet2 Abeline network interconnected to Qwest in the United States. The Abilene Network is a high-performance backbone that enables the development of advanced Internet applications and the deployment of leading-edge network services to Internet2 universities and research labs across the country. GEMnet provided Internet2 interconnection to the NTT labs in Musashino, Japan; KreoNet supplied Internet2 interconnection to the KT labs in Daejon, Korea.

Figure 1 illustrates the physical links between the four MPLS sites that make up the GMI 2004 test network. A combination of Alcatel and Cisco switches and routers established the core network at each site.

To create a 'normal' business environment, test engineers at each lab implemented a split-shift schedule. Each lab had two, six-hour shifts; each six-hour shift overlapped a shift at another lab and used the geographically dispersed equipment in each lab's time zone.

Three test equipment manufacturers participated in GMI 2004: Empirix, Navtel Communications, and Spirent Communications. They deployed their gear and test engineers at all four host sites. The hosts made their facilities available two weeks before the demonstration, so the network infrastructure could be built out and connectivity issues resolved before testing began.

Figure 1 **Core network topology**



Test Scenarios

Scenario 0: MPLS core

The basis for GMI 2004 was a traffic-engineered MPLS network designed in partnership with the MPLS and Frame Relay Alliance. Scenario 0 focused on quality of service (QoS) in the MPLS core, which is a key enabler for user-perceived voice quality during periods of congestion and oversubscription. Tests also examined the setup of MPLS traffic engineering (MPLS-TE) and EXP-inferred label switched paths (E-LSPs) in a heterogeneous environment.

It is essential to show that a network can deal with route failure and congestion. Redundancy is an essential requirement for high availability. Supporting route failover may be adequate in a non-latency-sensitive environment. However, when handling latency-sensitive real-time protocol (RTP) streams, the failover must happen almost instantaneously to help ensure an uninterrupted and undetectable failover for those using the network.

Within the GMI 2004 test network, MPLS protection was achieved using hot standby backup paths to protect single LSPs. This worked well.

Figure 2 **Scenario 0: MPLS core**

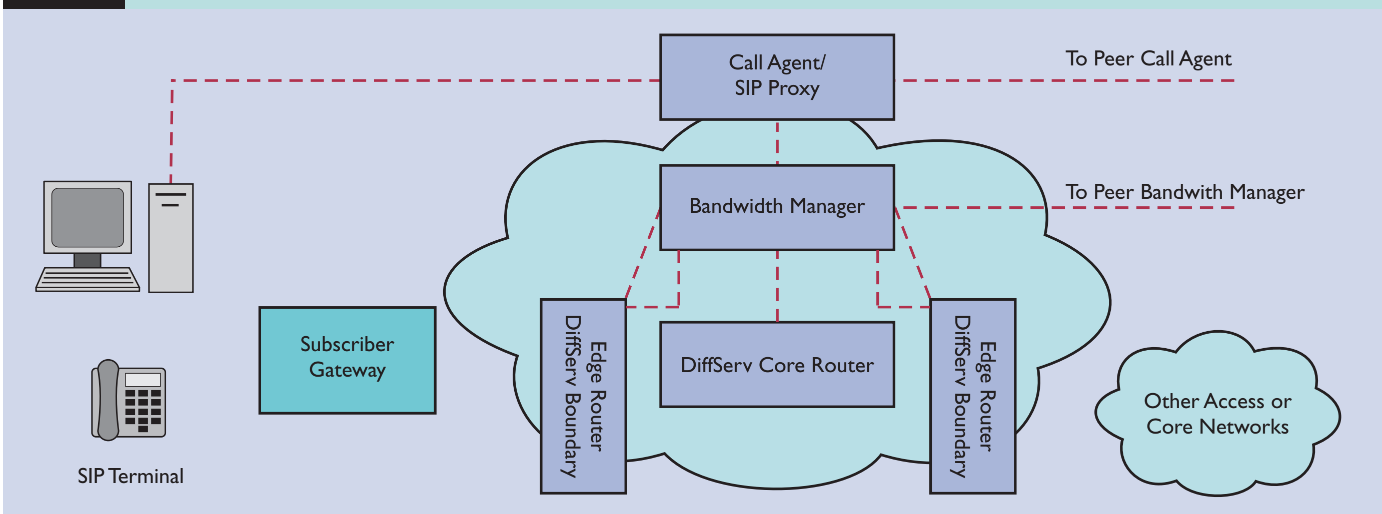
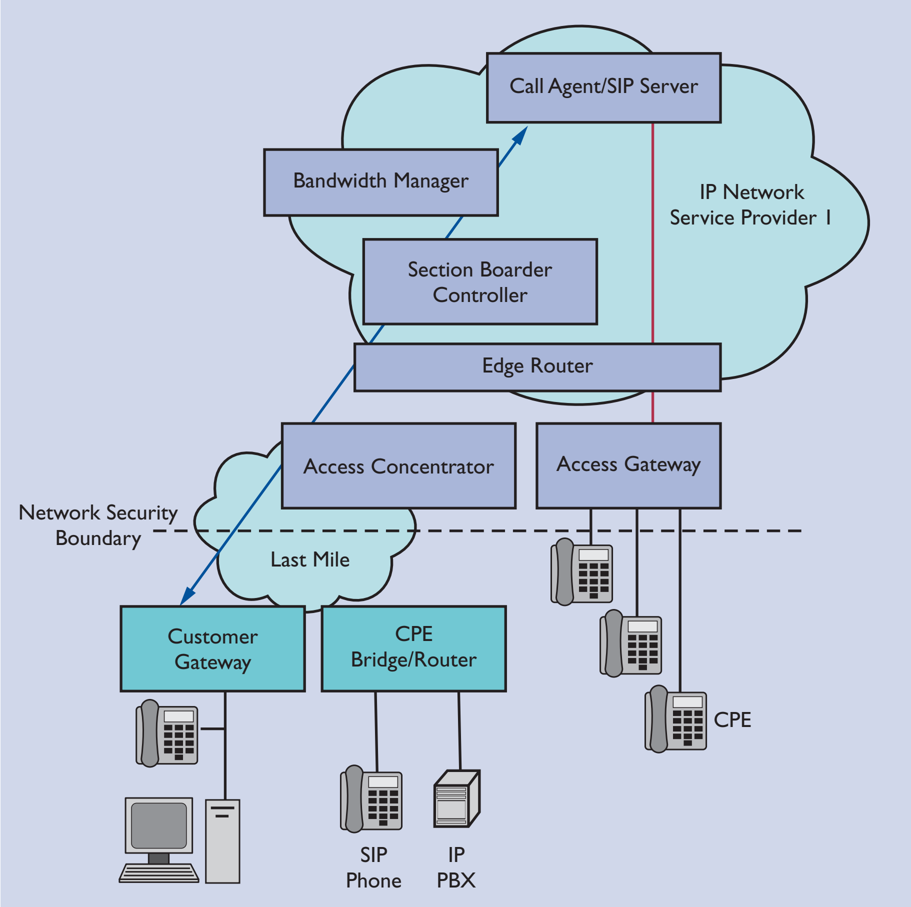


Figure 3 Scenario 1: single IP domain



In a larger global network, MPLS fast reroute (FRR) around a failed node or link would meet the needs of real-time applications, such as VoIP, that require user traffic to be redirected onto backup LSP tunnels in tens of milliseconds. This timing requirement is satisfied by computing and signalling backup LSP tunnels, in advance, and by redirecting traffic as close as possible to the point of failure.

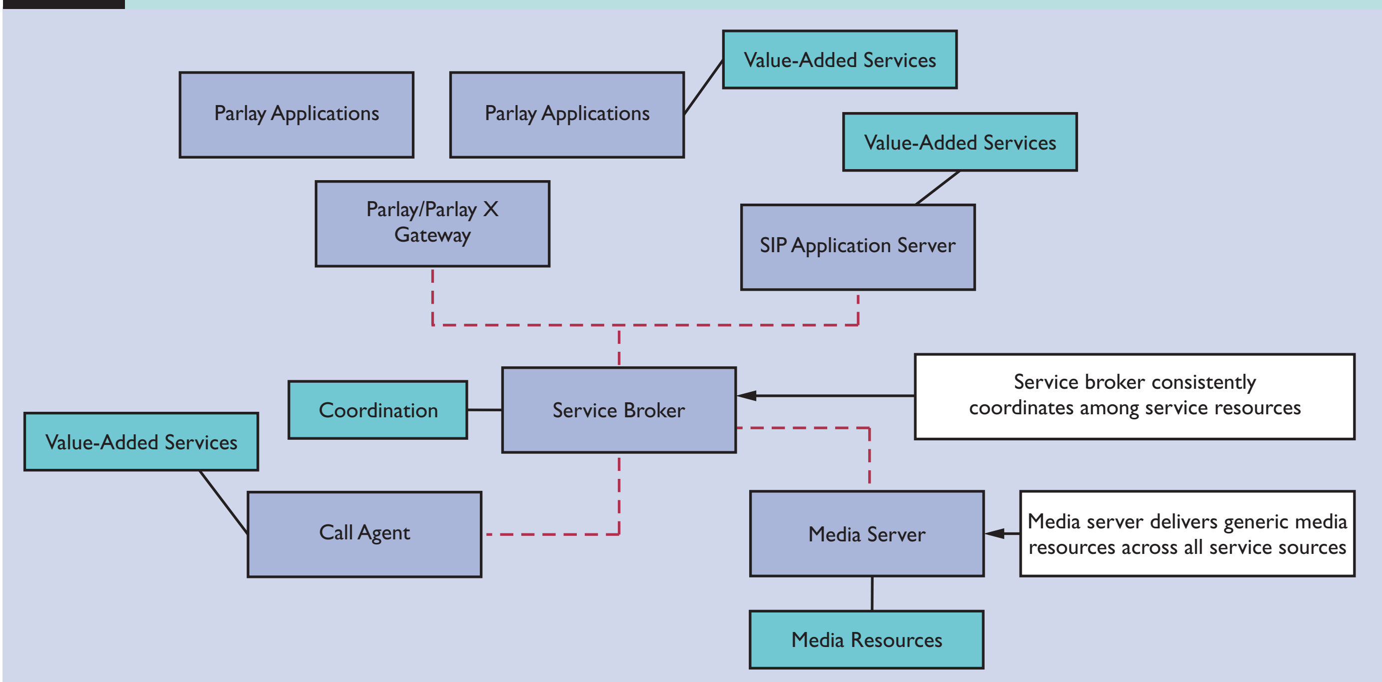
Scenario 1: Single IP domain

Scenario 1 served as the foundation for virtually all other tests conducted during the event. It involved a single call agent (CA)/SIP server within a single IP domain. This scenario involved more than 19 test cases exercising a multitude of basic IP telephony capabilities. Scenario 1 included the demonstration of several key aspects of the MSF QoS solution, including the ability of bandwidth managers to accept or reject calls based on tunnel/link occupancy and the ability of the underlying network infrastructure to provide QoS and protect traffic from network overload. Calls were set up between a variety of devices, including black phone to black phone, SIP phone to access gateway (black phone), and SIP to SIP phone. Protocols tested included H.248, media gateway control protocol (MGCP), and SIP. Basic consumer and business services such as call waiting, call forwarding, direct-dial in/direct-dial out (DDI/DDO), fax over IP, and modem over IP. This scenario demonstrated the basic features needed to show equivalence to today's traditional PSTN network.

Scenario 2: Single IP domain with value-added services

Building upon the basic capabilities tested in Scenario 1, Scenario 2 begins to layer on value-added services. The introduction of application servers, Parlay gateways, and service brokers allows providers and carriers to begin differentiating themselves from one another and from traditional PSTN offerings. The value-added services demonstrated in this scenario included OSA/Parlay enabled first-party and third-party priority calling, IP conferencing, click-to-connect/conference, and number translation (for example, VPN or FreePhone to PSTN). The value-added services in Scenario 2 leveraged the full

Figure 4 Scenario 2: single IP domain with value-added services



capabilities of the MSF architecture: The combination of SIP signalling, value-added services and bandwidth management available in Scenario 2 were used to demonstrate PIN collection and authorisation of priority calls, followed by successful completion of the priority calls despite severe network congestion.

Scenario 3: Single IP domain with value-added services and PSTN connectivity

Continuing to build on Scenario 2, Scenario 3 adds off-net calling to the PSTN. At this point, providers start to see how advanced features integrate with the legacy PSTN. By introducing application servers, parlay gateways, service brokers, and the PSTN, the bigger picture begins to form. Priority calling between SIP and SS7-based networks was demonstrated.

Scenario 4: Multiple IP domains and PSTN connectivity

In Scenario 4, calls were made across multiple IP domains, as well as to and from the PTSN. Services from earlier scenarios were used, and emergency call functions were extended to include continuous call retry and emergency call report/alarm.

Figure 5 Scenario 3: Single IP domain with value-added services and PSTN connectivity

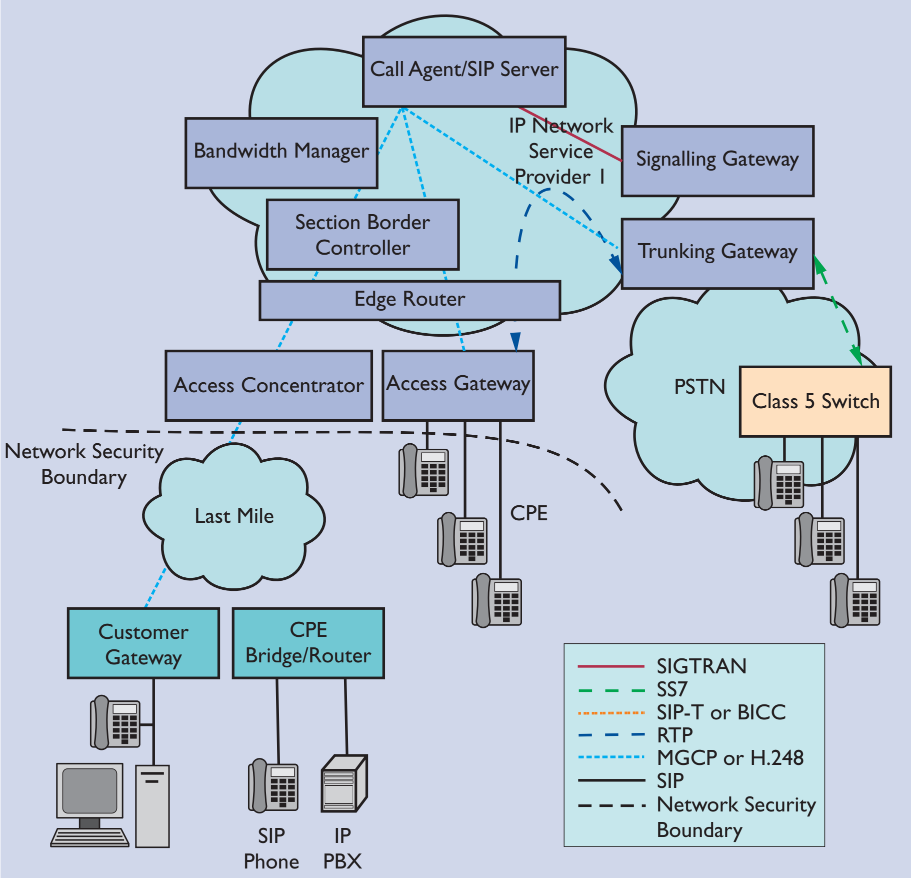


Figure 6 Scenario 4: multiple IP domains and PSTN connectivity

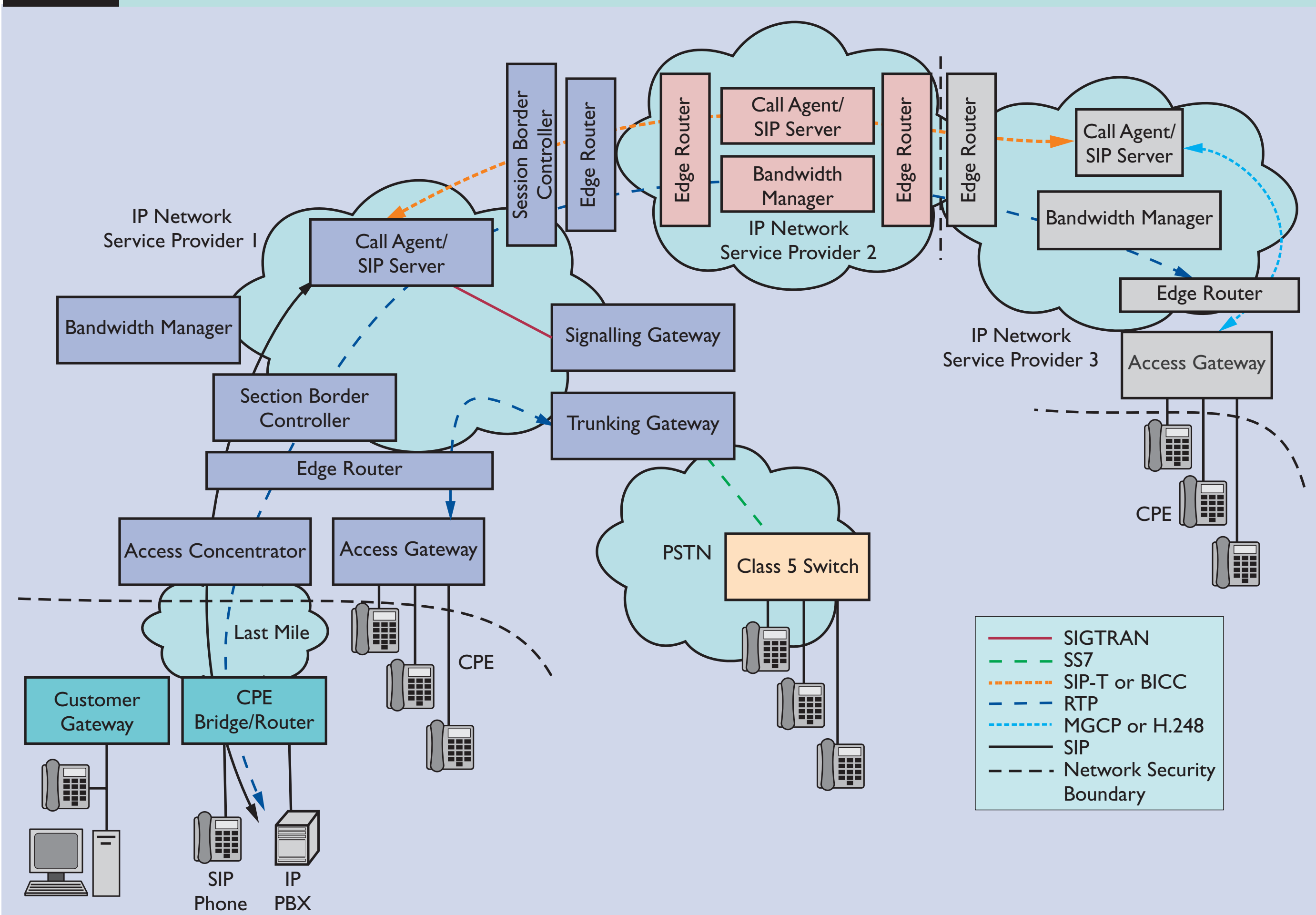
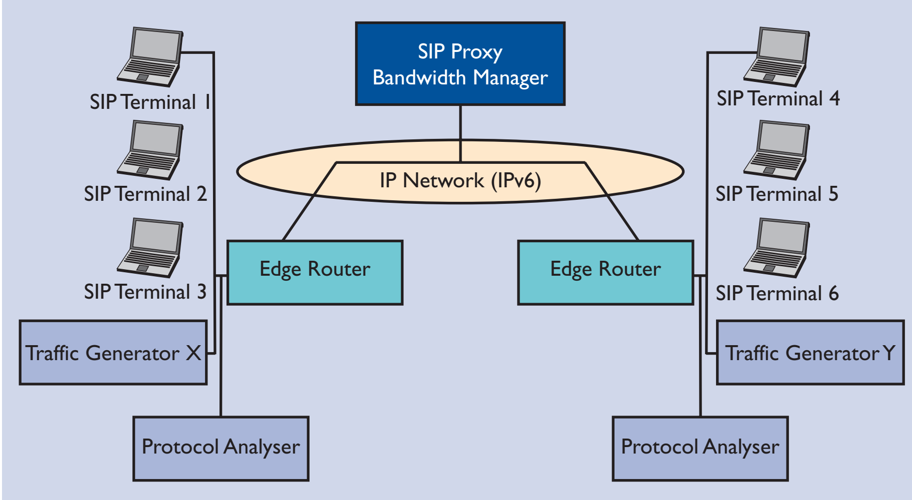


Figure 8 Scenario 6: IPv6 voice plus video



Scenario 5: Multiple IP domains with advanced services and PSTN connectivity

This scenario represents the final network configuration for GMI 2004. It demonstrated voice capabilities by adding the application environment on the access and egress network domains. Scenario 5 validated the MSF framework for global IP networks, delivering Parlay-enabled value-added services, such as priority calling, across multiple IP domains and to the PSTN.

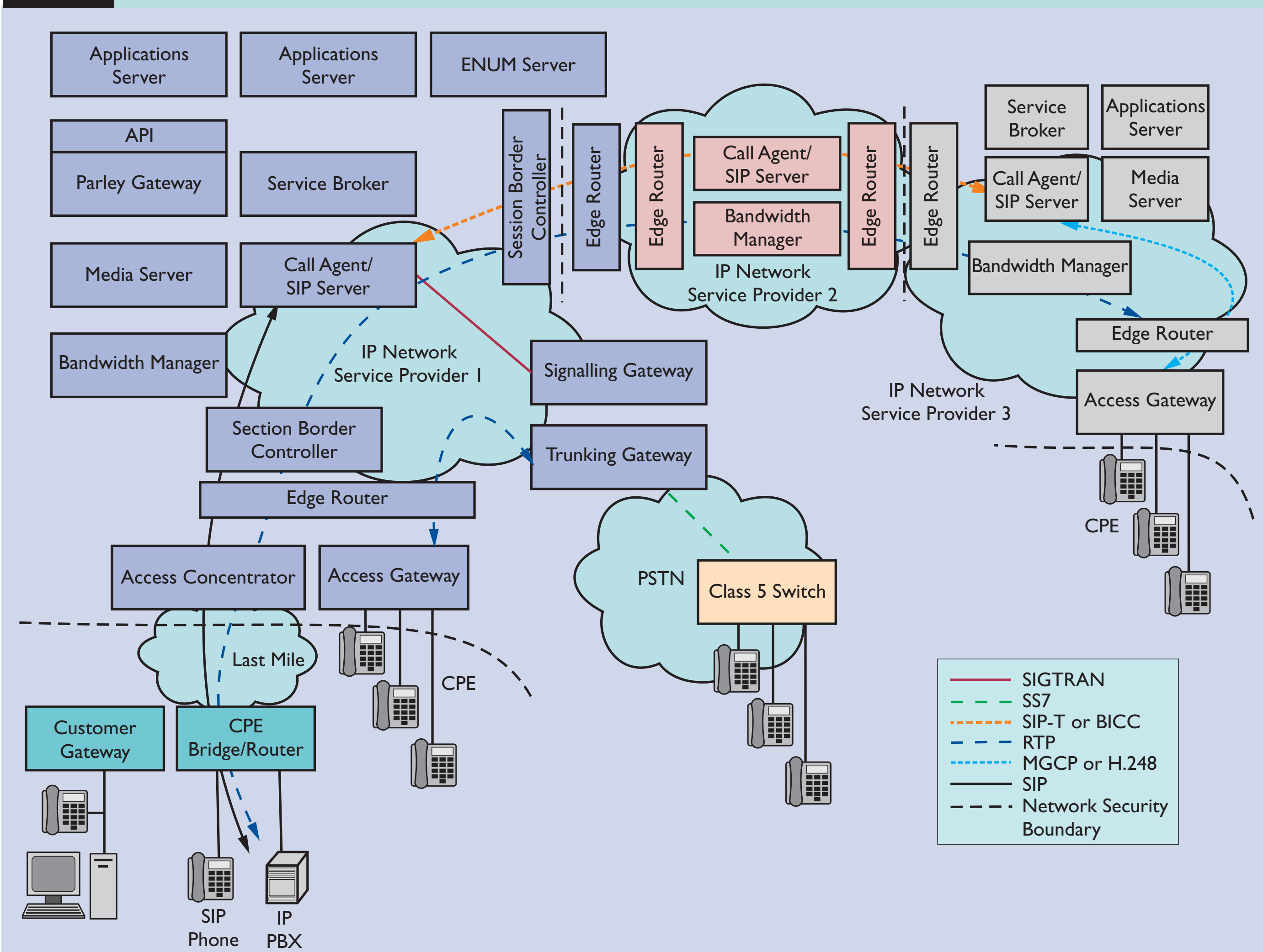
Scenario 6: IPv6 voice plus video

In collaboration with the North American IPv6 Task Force (NAv6TF) Moonv6 Project, Scenario 6 created an additional scenario. Tests implemented IPv6 from the NTT lab in Tokyo to the Moonv6 network in the United States at the University of New Hampshire InterOperability Laboratory (UNH-IOL). The network, enabled by Internet2, allowed several types of calls to be demonstrated. These included voice only, video only, and both video and voice; they were initiated and terminated with the SIP protocol. Further testing at the NTT Musashino Research and Development Center demonstrated QoS-controlled SIP voice and video calls.

Scenario 7: Management

Once a robust network is built, it must be managed. The MSF stipulates management solutions that are flexible and easy to work with, from provisioning through to maintenance. In collaboration with the TeleManagement Forum (TMF), the MSF developed a test scenario to monitor the MPLS core end-to-end, including the QoS pipes at each network site. This was the first time the MSF integrated management as a function of the global test network.

Figure 7 Scenario 5: multiple domains with advanced services and PSTN connectivity



Implementation Agreements

Table 1 lists the MSF IAs that were demonstrated during GMI 2004.

Results and Recommendations

Within each GMI 2004 scenario, test engineers attempted to evaluate as many combinations of vendor equipment for interoperability as possible. The overall success rate was 91 per cent. Table 2 gives the details for each scenario, as well as the average success rate for all seven scenarios.

GMI 2004 was designed to evaluate the accuracy and effectiveness of the MSF's Release 2 Architecture and IAs. As such, the event identified several issues that will form the basis for new MSF technical work. When interoperability issues were identified, the event monitors consulted test participants to accurately document discrepancies. In some instances, participants fixed the identified issues on the fly and were able to retest the scenarios and demonstrate interoperability. Those issues that could not be resolved were reported to the MSF's Technical Committee for review.

Scenario 0: GMI 2004 MPLS core network

There are several key points that should be noted for Scenario 0. The four host sites were treated as four separate areas. All host sites were running MPLS RSVP-TE signalling within the site. DiffServ is used for traffic classification within each site and between sites. Finally, all participants were expected to support the EXP field.

Scenario 1: Single call agent/SIP server within a single IP network service provider domain

Scenario 1 is the base scenario for GMI 2004; all other scenarios build on it. Table 3 illustrates basic components added to the core network to demonstrate Scenario 1. Table 4 shows the protocols used between each combination of access device.

Table 5 summarises the issues that were discovered in Scenario 1 tests.

Table 3 Components added to core network in Scenario 1

- SIP phone
- Black phone
- Call agent/SIP server
- Access gateway

Table 1 Implementation agreements demonstrated during GMI 2004

IA	Interoperating Devices
SIP	Application server/service broker
SIP	Service broker/call agent
MGCP	Call agent/trunking gateway
MGCP	Service broker/media server
SIP	Service broker/media server
COPS	Call agent/bandwidth manager
SIP	Call agent/bandwidth manager
MGCP	Call agent/user agent/access gateway
H.248	Call agent/access gateway
SIP	Call agent/user agent
SIP	Call agent/call agent
QoS	Bandwidth manager/core router
QoS (H.248 EMP)	Bandwidth manager/SBC/edge router
QoS (MSF-modified GCP)	Bandwidth manager/SBC/edge router
Media (SDP, RTP and codec negotiation, security)	N/A
ATM Trunking Gateway (H.248)	N/A
IP Trunking Gateway (H.248)	N/A
Signalling Gateway (Sigtran or proprietary)	N/A
IP Line-Side Access Gateway (H.248)	N/A
SIP-T	N/A

Key:
EMP: Extended Megaco package
SBC: Session border controller
SIP-T: Session initiation protocol for telephones
GCP: Gateway control protocol
SDP: Session description protocol
N/A: Not applicable

Table 2 Test scenario success rate

Scenario	Total Test Cases	Successfully Completed Test Cases	Per Cent Completed
Scenario 0	3	3	100%
Scenario 1	19	17	89%
Scenario 2	10	9	90%
Scenario 3	9	7	78%
Scenario 4	10	9	90%
Scenario 5	8	8	100%
Scenario 6	3	3	100%
Scenario 7	3	3	100%
Total	65	59	91%

Table 4 Protocols applied in Scenario 1

Connected Access Devices	Protocols
SIP phone/PBX to SIP phone/PBX	SIP to SIP
Black phone to SIP phone/PBX	MGCP to SIP
Black phone to black phone	MGCP to MGCP
Black phone to black phone	H.248 to H.248
Black phone to black phone	MGCP to H.248
Black phone to SIP phone	H.248 to SIP

Scenario 2: Single call agent/SIP server within a single IP network service provider's domain with value-added service

Scenario 2 adds value-added services to Scenario 1, using service brokers, media servers, Parlay application servers, and Parlay gateways. A session border controller provides network address translation (NAT). The following value-added services were evaluated: call screening, do not disturb, dual-tone multifrequency (DTMF) detection, find me, voicemail, and three-way calling.

Table 6 lists the components added to Scenario 1 to build Scenario 2.

As with Scenario 1, Scenario 2 uncovered several issues. Table 7 lists them and summarises the details of each.

Scenario 3: Call agent/SIP server to PSTN within a single IP network service provider's domain

Scenario 3 added PSTN interworking to Scenario 2; value-added services are still included.

Table 8 lists components added to the network in Scenario 3 in order to connect to the PSTN. Table 9 lists the issues discovered in Scenario 3.

Table 5 Summary of issues discovered in Scenario 1 tests

Issue	Details/Resolution
Tone package support between vendors	There was a problem in many of the tests that involved connecting a tone or announcement. The problem appears to be caused by one participant's AGW not supporting the tone package used by another vendor's CA. Progress was made between these participants, though retesting was not completed.
410 error	There was an issue when one participant initiated an INVITE message with another participant's IP and DN and a 403 error was received.
Voice unclear	With some intersite testing, the call flow was successful but the voice was not always heard.
403 error	With some intersite testing, a 410 error was recorded but later resolved with no further information as to the resolution.
Absence of ACK	There was an issue with one participant not sending an ACK. Participant is working to resolve. It is believed to be an implementation issue not an IA issue.
Call forward on busy	When a slight deviation from the test plan was made (the line was busy due to off-hook instead of establishing a call), a problem with one participant's CA was identified. The CA was sending an alert even though an off-hook signal had been received. It also was noticed that an AGW did not handle the error correctly. After discussion this was decided to be an implementation issue for both participants rather than an IA defect.
Three-way calling	There was a problem identified between one participant's CA and another's Softphone with midcall changes to the bearer (media) channel; for example, when connecting an announcement or three-party conference bridge. Occasional problems in this area also were seen with other equipment, so it may be worth checking that this is adequately covered in the IAs.
Fax to fax	There was an issue with fax calls between two participants: calls could be successfully established but in-band fax negotiation always failed. It is believed that the problem relates to a mismatch between the gateway and CA configuration but it was not possible to prove this in the time available. A similar problem occurred with modem calls.
Call agent failover	Following a shutdown of the worker side of the call agent, calls in speech remain, but a ringing call was lost when it was answered. This was not expected, but it was not possible to determine whether this is a problem with the CA or the SIP residential gateway. It is not thought to be an IA issue.

Key: AGW: Application gateway DN: Domain name

Table 6 Components added for Scenario 2 tests

- Service broker
- Media server
- Parlay application server/Parlay gateway
- Session border controller

Table 7 Summary of issues discovered in Scenario 2 tests

Issue	Details/Resolution
Call screening	It was observed that the service broker was stripping out SDP from 180 ringing messages and that one of the CAs did not support PRACK. That caused the first test for call-screening 2 to be unsuccessful.
Three-way conference (SIP)	There were reported issues in trying to get intersite calls to work between Qwest and KT. Though both parties claim not to have made any changes, calls began to work after several hours of unsuccessful attempts.
Three-way conference, mixed environment (SIP, H.248, MGCP)	One site spent 12 hours over the course of three days to get this call to work. The site was able to get three H.248 calls with relatively few problems; trying to get all three phones (MGCP, H.248, and SIP) to work took an excessively long time. Additionally, a lack of resources meant the site had to wait for a phone to become available. Other difficulties getting the MGCP phone to work were noted. According to a report, one participant did not correctly map MGCP to SIP.

Key: PRACK : Provisional acknowledgement

Table 8 Components added for Scenario 3 tests

- Call agent with SS7 capability
- Residential gateway/trunking gateway
- H.248 access gateway

Table 9 Summary of issues discovered in Scenario 3 tests

Issue	Details/Resolution
Test tool unable to hear speech path	Although the test was successfully completed, the event monitor was not able to listen to the speech path on the test tool. Analysis of the signalling identified an issue with connecting ring-tone on incoming calls. Despite this anomaly the participant in question was successful with both incoming and outgoing calls – via other participants' trunk gateways.

Table 10 Components added for Scenario 4 tests

- Black phone/residential gateway
- SIP phones
- Call agent/SIP server
- SIP proxy
- Session border controller

Scenario 4: Multidomain interworking

Scenario 4 adds VoIP interworking between different IP network service providers. For example, calls that originate on IP network 1, are transferred across IP network 2, and terminated on IP network 3. Value-added services (application servers, Parlay gateways, service brokers, and media servers) are removed in Scenario 4.

Scenario 4 repeats the tests performed in a single domain in Scenario 1, this time evaluating them across multiple domains. Scenario 4 also revalidates some of the features for calls to and from the PSTN that were tested in Scenario 3.

Table 10 illustrates the components typically added to the core network in Scenario 4; Table 11 lists the issues uncovered during testing, as well as detailing if and how they were resolved.

Table 12 Summary of issues discovered in Scenario 5 tests

Issue	Details/Resolution
Interdomain SIP-to-SIP call using click to connect (Parlay/OSA API)	There was a reported issue between SIP clients where the call set-up was successful, yet voice was not heard on the remote end.
Interdomain SIP-to-SIP call using click to connect (Parlay X API)	There was a reported issue involving the service broker. It was later verified that there was a misconfiguration of the dialled number. There was insufficient time to retest this configuration. At this point, it is not believed that there was any issue with the IA.
Conference call (SIP or H.248 via Parlay X gateway	There were reported issues with voice. In one test, the call set-up was successful and the signaling verified, but the voice was not heard on the softphone. Due to the lack of a microphone on the softphone, digits were dialled: It appears that the media server used may not be able to pass DTMF tones. This may be an area for further investigation in an IA. There also were reported issues with another pair of participants not passing voice even though the signalling was verified.

Key: API: Application programming interface OSA: Open services access

Scenario 5: Interdomain call agent/SIP server across multiple IP network service provider domains and to PSTN with value-added services

Scenario 5 restores value-added services (service brokers, media servers, application servers, and Parlay servers) to the multidomain test environment of Scenario 4. Issues discovered during Scenario 5 testing are detailed in Table 12.

Summary

Call agents' lack of support for the stream control transmission protocol (STCP) was identified as a significant interoperability issue. Although the original IA (MSF-IA-

Table 11 Summary of issues discovered in Scenario 4 tests

Issue	Details/Resolution
Number Addressing of SIP calls	A number of tests either failed or could not be run because some equipment only accepts numbers in specific formats. For example, one call agent required '00' to be added to the front of international numbers, while one participant's application server always removes the international prefix and replaces it with a plus sign ('+'). Similarly, some equipment could only accept IP addresses, although this may have been due to the lack of DNS at some sites.
ACKs	There was a problem with ACKs between two participants. Although this does not appear to be IA-related, it could be related to the SIP routing issues discovered later in the test. There also was an issue between two participants talking via the SBC; no ACKs were received for the BYEs.
Unexpected IP address misconfiguration	One participant had a problem talking to another that was filtering out its messages because they had an unexpected IP address combination. This is believed to be a configuration issue rather than an IA problem.
Bandwidth Manager/ SDPproblems/ one-way speech	One participant first had problems working via the bandwidth manager and then experienced trouble with SDP. Once these issues were resolved, calls were possible, but only one-way speech. This is believed to be a vendor implementation issue rather than an IA issue.
Dial-plan errors	When the microphone was missing for one of the soft clients, dialling numbers were used instead of voice to validate a successful call. There were reported issues with the dial plan for international calls. One participant's SIP proxy stripped the first four digits (0 and country code); if the device on the other end was looking for all of the digits, it had to be configured so that it accepted the number without the first four digits. Further investigation is needed to determine if this issue should be addressed in an IA.
Call transfer	At one of the sites, it was determined that only one setup worked: two MGCP phones. A participant attempted to call from an MGCP to SIP, flash another SIP agent, and then hang up the MGCP. This call failed because the originating CA did not send SDP in the invite, and the terminating CA did not send SDP in its 200 OK (request successful) responses to the INVITE.
CLI validation	There were reported issues when SBC outbound calls were successful, while inbound calls failed. It also was noted that a remote SBC was sending calls directly to the local SBC. This caused a 200 OK not to receive an ACK from the local site, which caused it to release the call after the remote site's 200 OK retransmits and timeouts. The BYE also was missing a 200 OK. Further understanding of the SBC and its role in the network is needed. A possible refinement of IAs to better integrate SBCs may be necessary.
3-way calling	Similar issues were reported. Outbound calls via the SBC were successful while inbound calls failed. It also was noticed that when the local site called the remote site, there were ACK issues. The remote SBC was sending the local SBC an INVITE and ACK, when it should be sending them to the CA. Again, further understanding of the SBC and its role in the network is needed. Possible refinement of IAs to better integrate SBCs may be necessary.

Key: CLI: Calling line identity DNS: Domain name server

MEGACO.003-FINAL) shows SCTP support as mandatory for all call agents, only one participant supported SCTP. This protocol is more appropriate than user datagram protocol (UDP) for large trunk gateways in a replacement PSTN environment; thus it should be reinstated as mandatory for all MSF-compliant call agents controlling such gateways. The lack of SCTP on the call agents was not identifiable from the equipment list information available before the event, thus preventing remedial action or withdrawal from the event. Procedures should be introduced in future events to enable early identification of such major option choices that can completely preclude interworking.

Other interoperability issues identified during the testing include:

1 One participant used a periodic re-Invite to detect loss of an SIP end point during a call in an attempt to minimise any overcharging. This caused problems for several other participants, however. If network operators wish to use duration-based charging on calls from SIP end points, then further work needs to be done to introduce a standard mechanism that will be supported by all vendors.

2 There is a similar issue with another participant's H.248 trunking gateway which uses periodic service changes containing '999 HeartBeat' to check its connection to the call agent. Some consideration should be given to whether this mechanism would be acceptable in a large network or whether another mechanism would be better (for instance, SCTP instead of UDP).

3 When using SIP via an SBC or bandwidth manager, further clarification is needed on SIP routing and usage of SIP headers to ensure that the required messages all go via the SBC/BM in both directions. This caused a certain amount of difficulty when initially configuring the network, especially when DNS was used to resolve SIP uniform resource identifiers (URIs).

4 Questions were raised about whether a service broker or application server is allowed to change the To and From headers in SIP messages.

5 There were a number of tests that either failed or could not be run due to various items of equipment only accepting numbers in certain formats. In one example, a participant's call agent required '00' added at the front of international numbers, but another participant's application server always substituted a '+' the international dialing prefix. To ensure true interoperability, further work is required on the addressing of SIP calls (and the resolution of those addresses) if they are to be used across and between carrier-scale networks.

6 A number of vendors appeared to have problems interworking between H.248 versions. If no guidance on backward (and forward) compatibility is included in the specification or MSF IA, then it may be worth including some.

7 There seemed to be a lot of issues and discussions about H.248 end-point naming. This may be an area worth checking whether the IA needs expanding in this area.

8 There also were a number of discussions about capitalisation in H.248. For example should it be 'ROOT', 'root' or 'Root'? The problems were resolved during the event by a participant changing its code, but to ensure long-term interoperability it may be worth emphasising the MSF requirements in the IA.

9 Initially some problems were encountered with call agents trying to download H.248 digit maps that were too big for a gateway to handle. This was resolved for the event by simplifying the digit maps in the call agent configuration. If the MSF wants to deliver true 'plug and play' interoperability, then the IA should specify the maximum size digit map the CA can send (or a minimum size the gateway must accept) and, if possible, some mechanism to negotiate up from this.

10 A participant raised an issue about how early media cut-through should be achieved on SIP calls involving an application server (or service broker) when this is required by a service.

11 SIP error handling seemed lackluster at best. SIP error messaging seemed to vary from just giving up and returning to idle without attempting to tell the other party what had happened to sending lots of BYEs or in an attempt to reset everything to idle. This may be another area in which some guidance in the IA could be useful.

12 Gateways sending repeated Session Close (Forced) (SC Forced) messages may not be the best method to handle a close. Creating an IA to stop this might warrant some thought and discussion.

13 The ability in H.248 to send an error message without a transaction should be investigated further, and possibly an IA should mandate the use of a Reply = Transaction with an embedded error descriptor.

14 The hiG media gateway sent a TransactionResponseAck message appended to an existing transaction. This seems to be of little value as some equipment might dump this via a preparsing step.

15 Further work needs to be accomplished in the MSF regarding SIP proxy-based SBCs. Issues such as which headers to change and how to perform routing were discovered. Calls were possible thanks mainly to the bandwidth manager being able to apply routing rules over and above normal SIP. Related to this is the general issue of how SIP messages are routed in a 'real' network, which has lots of proxies and SBCs and seems to have to rely on a combination of DNS and applied rules at points along the way.

16 Thought should be given to having the IAs stipulate something about being able to handle both domain names and IP addresses in the host part. Implementations were seen that did one or the other, forcing the bandwidth manager to apply a rule to get two devices to be able to talk to each other.

17 Investigation of the use and implementation of DNS in a large-scale VoIP network is warranted.

18 One key result was noting the message storm that resulted whenever a SIP call went wrong. This was due to stacks not obeying the recommendations in the core SIP IA with retry timers and numbers of retries.

19 Not strictly an IA issue, but SIP-based networks need extensive proving to ensure that messages do not get sent to the wrong place. Various issues were found with strict/loose routing and headers not being popped. Sometimes, this has bizarre results: For example, an ACK was observed 17 seconds after the 200 OK.

20 The inability of SIP stacks to handle midcall bearer via UPDATE reflects the fact that most stacks haven't been crafted according to the IAs.

Conclusion

With the growing competition for voice services, incumbents are being forced to look closely at the maturing packet-voice technology and advanced services that go along with it. Making such a radical change from the traditional PSTN must be done cautiously, however. Ensuring interoperability is a key aspect that must not be overlooked. The migration to a packet-based telephony network will not happen overnight. Hence, the work of the MSF is critical to ensure concurrent and fully interoperable packet and legacy networks.

GMI 2004 brought together experts from vendors and carriers to highlight and address protocol ambiguities and offer a more clearly defined path to implementation. Over the 12 days of GMI 2004, 100 engineers worked 14 hours a day, resulting in 16 800 man-hours of work. The IAs tested were created to address real issues facing the global carrier community, ranging from multivendor interoperability to value-added services to priority calls to QoS management.

During the testing, protocol issues were resolved in a collaborative way and documented. These issues will be passed to the proper standards bodies and addressed in future MSF IAs. The goal was successfully demonstrated with one robust next-generation network in GMI 2004.

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Biography

Roger Ward
BT

Roger Ward has been heavily involved in the MSF from its foundation in 1998. In June 1999 he was elected to the MSF Board of Directors and, in September 2000, he was elected MSF President. In his work with the MSF, he has done much to help achieve the forum's current focus on multi-vendor interoperability and the very successful GMI 2002 and GMI 2004 global interoperability programmes. Based in the UK, he started his BT career in 1974 after graduating with an honours degree in Electrical Engineering from Cambridge University. Since then he has had a long distinguished career in the industry holding a variety of key positions. In the 1980s, he had design authority responsibility for key aspects of both System X development, BT's local exchange competitive procurement programme ('System Y'), early deployment of Centrex in the UK and the derived services network (DDSN). During the 1990s he was responsible for advanced platforms and intelligence strategy, working with MCI on the merger project which led to BT's next-generation switch deployment in the UK trunk network. He holds an M.Sc. degree in Telecommunication Systems from Essex University and an MBA from Warwick University, the latter being achieved after being selected by BT to participate in Warwick Business School's inaugural Integrated MBA programme. He is a Chartered Engineer and a Member of the Institute of Electrical Engineers, and currently heads up BT's involvement with industry fora within the BT Group Technology Office.