MSF Global Interop (GMI) 2004
Physical Test Scenarios
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Multiservice Switching Forum Technical Report

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Abstract:

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 this end the MSF looks to adopt pragmatic solutions in order to maximize the chances for early deployment in real world networks.

To date the MSF has defined a number of detailed Implementation Agreements and detailed Test Plans for the signaling protocols between network components and is developing 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.

In 2002, the MSF held a “Global MSF Interoperability 2002” (GMI 2002) event that tested interoperability between next generation network elements situated in Asia, Europe and North America. GMI 2002 validated the MSF release 1 architectural framework and Implementation Agreements by subjecting them to interoperability testing based on realistic network scenarios.

Following the success of GMI 2002 the MSF work program continues to address the key technical barriers to next generation network deployments. Global MSF Interoperability 2004 (GMI 2004) will demonstrate a deployable and operationally ready IP telephony network with Network Management, enhanced Quality-of-Service (QoS) and security features. GMI2004 will also demonstrate a service layer with application server, media server, and service broker functionality. This will enable the MSF to demonstrate a full end-to-end customer ready deployable network.

It is envisaged that GMI2004 will provide an industry showcase that will:

- Assist carriers achieve their goal: to deploy flexible, best of breed products.
- Assist vendors achieve their goal: to market products more cost effectively.
- Display the global interoperability of the MSF architecture as referenced in the Release 2 architecture document.
- Demonstrate a network scenario that can be managed to specific quality standards.

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/

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1 Introduction

Global MSF Interoperability 2004 (GMI 2004) will demonstrate a deployable and operationally ready IP telephony network with Network Management, enhanced Quality-of-Service (QoS) and security features. GMI2004 will also demonstrate a service layer with application server, media server, and service broker functionality. This will enable the MSF to demonstrate a full end- to- end customer ready deployable network.

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- Assist carriers achieve their goal: to deploy flexible, best of breed products.
- Assist vendors achieve their goal: to market products more cost effectively.
- Display the global interoperability of the MSF architecture as referenced in the Release 2 architecture document.
- Demonstrate a network scenario that can be managed to specific quality standards.

This document contains the GMI 2004 Physical Scenarios that will be used to derive test plans and test setups for the GMI 2004 event.

GMI 2004 will use 5 physical scenarios that build upon one another. The scenarios are

- Scenario 1 - Single Call Agent / SIP Server within a single IP Network SP domain.
  This is the base scenario for GMI 2004.

- Scenario 2 - Single Call Agent / SIP Server within a single IP Network SP domain with Value Added Services.
  This scenario adds Value Added Services to scenario 1 using Service Brokers, Media Servers, Application Servers and Parlay Servers.

- Scenario 3 - Call Agent / SIP Server to PSTN within a single IP Network SP domain.
  This scenario adds PSTN interworking to scenario 2. Value Added Services are still included

- Scenario 4 - Call Agent / SIP Server across multiple IP Network Service Provider domains and to PSTN.
  This scenario adds VoIP interworking between IP Network Service Provider domains, but does not include Value Added Services.

- Scenario 5 - Call Agent / SIP Server across multiple IP Network Service Provider domains and to PSTN with Value Added Services.
  This scenario adds Value Added Services to scenario 4 using Service Brokers, Media Servers, Application Servers and Parlay Servers.

This document does not include the management scenarios for GMI 2004 which are documented separately and based around these 5 physical scenarios.

1.1 Background

This set of 5 scenarios is based upon contributions made to the Vancouver meeting, discussions at the meeting and follow up material provided at or shortly after the meeting.

The following table indicates which contributions were used to derive these 5 scenarios.

Scenario 1	msf2003.100.00	GMI2004 Service scenario for IP PBX & Access Gateway	Adrian Sapwell Mike Bick
	msf2003.102.00	GMI2004 Service Scenario Class 5 Softswitch/SIP Server to PSTN	Wayne Cutler
Scenario 2	msf2003.099.01	Draft GMI2004 Value Added Service Scenario	Chris Daniel

	msf2003.110.00	GMI 2004 Parlay ETS-Services Demonstration: Scenario and Physical Network Framework	John Wullert
Scenario 3	msf2003.101.00	GMI2004 Service scenario for Access Gateway & Class 5 switch	Adrian Sapwell Mike Bick
	msf2003.115.00	Proposed GMI 2004 Demonstration - IEPS Priority Call Setup Message over an IP Bridge	Frank Suraci Jack Garrity
Scenario 4	msf2003.090.00	RFN Samples	Mike Kallas Mitch Laman
	msf2003.102.00	GMI2004 Service Scenario Class 5 Softswitch/SIP Server to PSTN	Wayne Cutler
Scenario 5	msf2003.099.01	Draft GMI2004 Value Added Service Scenario	Chris Daniel
	msf2003.110.00	GMI 2004 Parlay ETS-Services Demonstration: Scenario and Physical Network Framework	John Wullert

2 VoIP Next-Generation Network Architecture

The overall end to end Network Architecture for GMI 2004 is shown in figure 1 below.

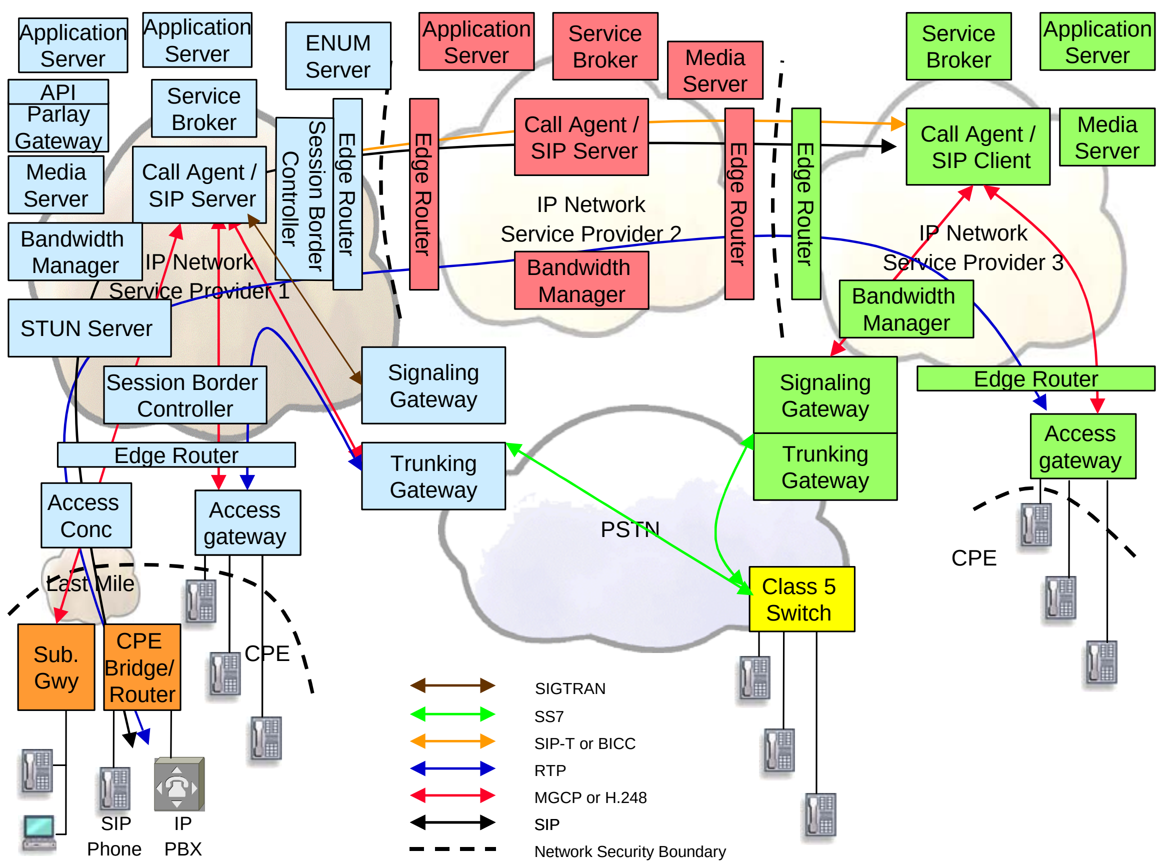

Figure 1: Next Generation VoIP Network

The example VoIP (IPv4) Next Generation network contains 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 or SIP-T 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.

Each of the physical scenarios is based upon this overall end-end architecture and tests a subset of the overall solution.

3 Scenario 1 - Single Call Agent / SIP Server within a single IP Network SP domain

This is the base scenario for GMI 2004 and is shown in figure 2 below.

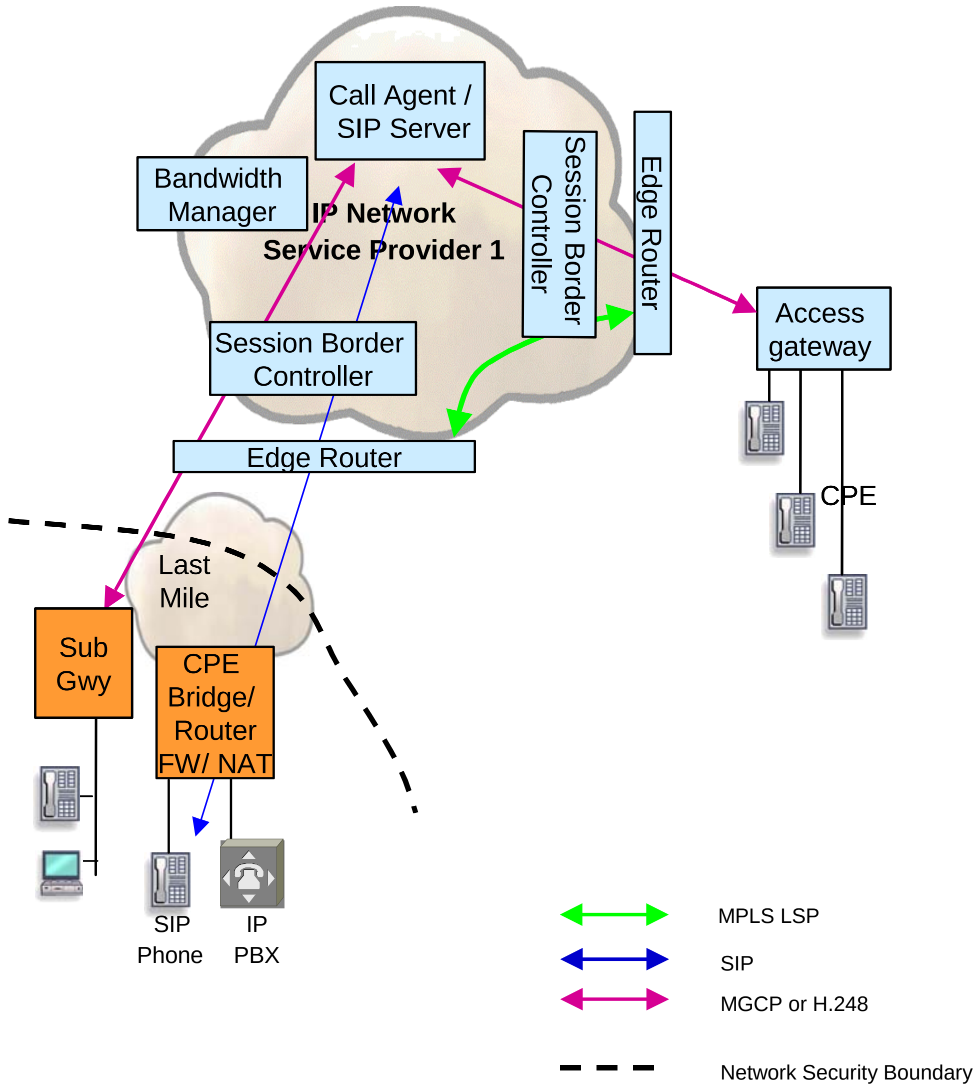

Figure 2: Scenario 1 Physical Architecture

3.1 Domains

This scenario is limited to a single IP network within a single service provider domain. The domain can either run in a public or private IP address space.

3.2 Network Components

The following components are included in Scenario 1

- One redundant Call Agent / SIP Server
- One redundant Bandwidth Manager
- One or more redundant Session Border Controllers
- At least two Edge Routers

- At least two Access Devices which can be of the following types
    - Access Gateway
    - IP Phone
    - IP PBX
    - Subscriber (Residential) Gateway (IAD)
- One or more CPE Firewall/NAT device

3.3 Network Links

3.3.1 Last Mile

For IP Phones, IP PBX and Subscriber Gateway the last mile network is IP over a broadband access connection, e.g. T1/E1, DSL, Cable, Metro Ethernet, Fixed Wireless, FTTH.

For Access Gateways the last mile network is either analog POTS or ISDN (PRI or BRI).

3.3.2 Core IP Network

The core IP network between edge routers will be running over MPLS Label Switch Paths (LSPs), although non-MPLS implementations for example ATM or Ethernet can also be included.

Components such as the Call Agent / SIP Server are expected to be connected via Ethernet, but can also use ATM, MPLS or other layer 2 protocol.

3.4 NAT / Firewall

Network Address Translation (NAT) and Firewall function is included at the customer premise. In particular IP Phones, IP PBXs and Subscriber Gateways can reside behind 'standard' NAT and firewalls.

This scenario uses either of the following solutions to cope with NAT and Firewall.

- A Session Border Controller which sits on both the media and signaling path.
  For SIP the Session Border Controller acts as a B2BUA and both translates the IP addresses in the SIP signaling and redirects the RTP traffic.
  For MGCP the Session Border Controller acts as both an MGC and a MG and both translates the IP addresses in the MGCP signaling and redirects the RTP traffic.
- A firewall/RTP proxy device controlled via one of the protocols listed in section 3.7.

3.5 Security

This scenario includes the following security solutions for SIP Phones and SIP PBXs

- HTTP Digest for authentication
- TLS for encryption and Integrity

There are no security solutions included for MGCP or H.248 controlled Access Devices.

3.6 Quality of Service

The overall MSF QoS solution is outlined in the Quality of Service for next generation VoIP networks solution framework (msf2003.105.00).

In particular the following are included in scenario 1.

- In the core IP Network two or more MPLS LS Edge Routers controlled by a Bandwidth Manager using traffic priority.
- QoS enabled back-haul based on Admission Control (no dial tone/ equipment busy).
- In the access network, DiffServ or an alternative prioritization marking such as Ethernet 802.1p used to mark voice and signaling traffic and prioritize over data traffic.

3.7 Protocols and Implementation Agreements

The following protocols and Implementation Agreements are used in scenario 1.

Note: where the IAs are not yet published the current draft contribution is referenced in parentheses.

Interface	Protocol	Implementation Agreement	MSF IA (Draft Contribution)
Call Agent <-> Access Gateway	H.248	Implementation Agreement for MEGACO/H.248 Profile for an IP Line Side Access Gateway.	MSF-IA-MEGACO.004-FINAL (msf2002.198)
	H.248 & IUA for UK	Multi Service Access Gateway Implementation Agreement	MSF-IA-MEGACO.005-FINAL (msf2003.117)
	MGCP	Implementation agreement for MGCP for voice over IP between a call agent and a user agent	MSF-IA-MGCP.001-FINAL
Call Agent <-> Subscriber Gateway	MGCP	Implementation agreement for MGCP for voice over IP between a call agent and a user agent	MSF-IA-MGCP.001-FINAL
Call Agent <-> IP Phone or IP PBX	SIP	SIP IA - CA-UA	MSF-IA-SIP.003-FINAL (msf2003.071)
Call Agent <-> Bandwidth Manager (IF-2)	SIP	SIP IA - CA-BM	MSF-IA-SIP.010-FINAL (msf2004.027)
	DRIP	Implementation Agreement for a Dynamic Resource Initiation Protocol (DRIP)	MSF-IA-DRIP.001-FINAL (msf2003.066)
Bandwidth Manager <-> Edge Router (IF-3)	Not Required	N/A	N/A
	ETSI H.248 EMP	ETSI Tiphon EMP package	MSF-IA-MEGACO.006-FINAL (msf2004.026)
	GCP	Implementation Agreement for MSF Variant Gate Control Protocol	MSF-IA-GCP.001-FINAL (msf2003.114)
	Pinhole Control Protocol	RTP Proxy/FW control protocol	msf2003.113
Bandwidth Manager <-> Core Router (IF-4)	SNMP	Proprietary management interface using SNMP	N/A
	CLI	Proprietary management interface using command line interface	N/A
Call Agent <-> Session Border Controller (RTP Proxy/FW)	Not Required	N/A	N/A
	Pinhole Control Protocol	RTP Proxy/FW control protocol	msf2003.113

	ETSI H.248 EMP	ETSI Tiphon EMP package	MSF-IA-EMP.001-FINAL (msf2003.022)
	GCP	Implementation Agreement for MSF Variant Gate Control Protocol	MSF-IA-GCP.001-FINAL (msf2003.114)

The following IAs are required by several of the IAs listed in the above table

- Implementation Agreement for CORE SIP Profile, for Voice over IP,
  MSF-IA-SIP.002-FINAL (msf2003.132)
  Required by all the SIP IAs
- Implementation Agreement for SDP Usage & Codec Negotiation for GMI 2004,
  MSF-IA-SDP.001-FINAL (msf2003.059)
  Required by all the IAs which use SDP, which include MGCP, H.248 and SIP IAs.
- Implementation Agreement for SIP Signalling Security for GMI 2004,
  MSF-IA-SIP.008-FINAL (msf2003.116)
  Required for SIP IA – CA-UA, i.e. by SIP Phones and PBXs and Call Agents.
- Implementation Agreement for MGCP VoIP between CA and UA – Security Addendum,
  MSF-IA-MGCP.002-FINAL (msf2004.008)
  Optional support for MGCP between CA and UA.

There are several supported models for providing NAT and Session Border Control function

- A standalone Session Border Controller monitoring MGCP/SIP and RTP traffic flowing through and not controlled via a specific protocol.
- An RTP Proxy/Firewall, which can be incorporated into an edge router controlled using one of the following protocols
  - Pinhole Control Protocol
  - ETSI H.248 EMP
  - MSF Variant Gate Control Protocol

3.8 Redundancy and Failover Requirements

The following redundancy and failover requirements are included in scenario 1

- Redundant Call Agents
  - 1:1 warm back up or single fault tolerant Call Agent with no single point of failure
  - Stable call preserved during call agent swap or internal failover
- Redundant SBC
  - (n+m) load sharing
  - Calls dropped during SBC swap
- Redundant Bandwidth Manager
  - 1:1 back up or single fault tolerant Bandwidth Manager with no single point of failure
  - Bandwidth state kept during bandwidth manager swap or internal failover
- Redundant links
  - To Call Agent
  - Within MPLS core network (MPLS fast re-route)

3.9 Feature List

The following features and combinations of access devices are included in scenario 1

- Basic Call with the following combinations
  - Access Gateway - Access Gateway (black phone - black phone)
  - Access Gateway - Access Gateway (fax - fax)
  - Access Gateway - Access Gateway (V.90 modem - V.90 modem)
  - IP PBX - Access Gateway (black phone)
  - SIP Phone – Access Gateway (black phone)
  - SIP Phone – SIP Phone
  - Subscriber Gateway - Access Gateway (black phone - black phone)
  - Subscriber Gateway - Access Gateway (fax - fax)
  - Subscriber Gateway - Access Gateway (V.90 modem - V.90 modem)
  - Subscriber Gateway (black phone) – SIP Phone
  - IP PBX – SIP phone
  - IP PBX - Subscriber Gateway (black phone)
  - IP PBX – IP PBX
  - Subscriber Gateway - Subscriber Gateway (black phone - black phone)
  - Subscriber Gateway - Subscriber Gateway (fax - fax)
  - Subscriber Gateway - Subscriber Gateway (V.90 modem - V.90 modem

- Support for Residential Features
  - Call Transfer
  - Call Waiting (Incoming call indicated using an in-band tone but no display of Calling Number or Name)
  - Call Diversion (Call Forwarding)
  - Calling Number and Name Delivery
  - Three-party conference (3-way calling)

- Support for Business Features
  - Centrex
  - Private numbering plan
  - DDI / DDO

3.10 Exception Testing

The following exceptions tests are included in scenario 1.

- Rejection of Calls when MPLS LSP bandwidth exhausted
- Load network with mix of Voice and Data traffic and check to see that voice quality is not impaired.
- Signalling load testing to check the Call Agent gracefully rejects calls when load exceeds maximum call rate and concurrent call capacity.

4 Scenario 2 - Single Call Agent / SIP Server within a single IP Network SP domain with Value Added Services

Scenario 2 builds on scenario 1 to add Value Added Services as shown in figure 3 below.

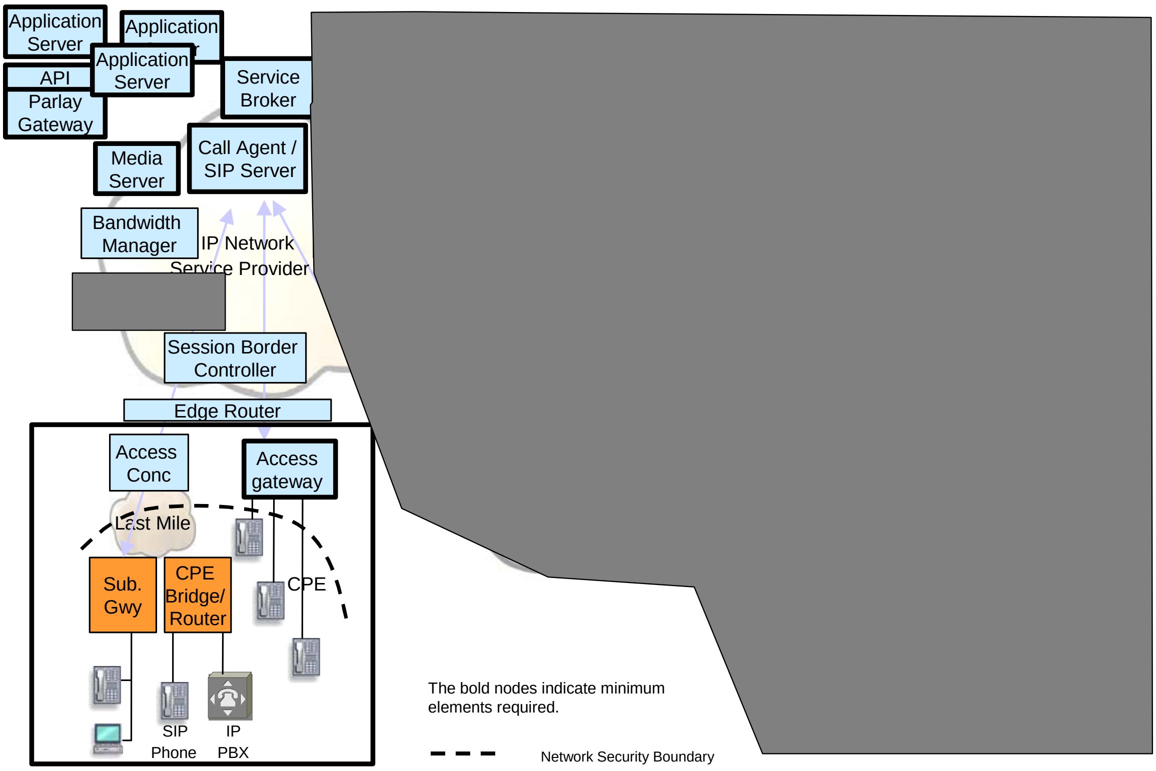

Figure 3: Scenario 2 Physical Architecture

The following sections add requirements and features to Scenario 1. Each section notes where the requirements of scenario 1 apply without change.

4.1 Domains

As for scenario 1.

4.2 Network Components

In addition to the components listed in scenario 1, the following components are added in scenario 2

- One or more redundant Application Servers.
- One or more redundant Parlay Gateways and Application Servers.
- One or more redundant Service Brokers.
- One or more redundant Media Servers.

Some of these network components can be combined, for example Media Server and Application Server or Service Broker and Call Agent.

For the remainder of this section the term Application Servers includes Parlay Gateways.

4.3 Network Links

4.3.1 Last Mile

As for scenario 1.

4.3.2 Core IP Network

As for scenario 1 with the following addition.

Components such as the Service Broker, Application Server and Media Server are expected to be connected via Ethernet, but can also use ATM, MPLS or other layer 2 protocol.

4.4 NAT / Firewall

As for scenario 1.

4.5 Security

As for scenario 1 with the following addition.

No security is required for Service Brokers, Application Servers or Media Servers.

4.6 Quality of Service

As for scenario 1.

4.7 Protocols

As for scenario 1 with the following additions.
Note: where the IAs are not yet published the current draft contribution is referenced in parentheses.

Interface	**Protocol**	**Implementation Agreement**	**MSF IA (Draft Contribution)**
Call Agent <-> Service Broker	SIP	SIP IA Call Agent to Service Broker	MSF-IA-SIP.005-FINAL (msf2003.064)
Service Broker <-> Application Server	SIP	SIP IA Service Broker to Application Server	MSF-IA-SIP.006-FINAL (msf2003.063)
Service Broker <-> Media Server	SIP	SIP Media Server IA	MSF-IA-SIP.009-FINAL (msf2004.006)
Service Broker <-> Parlay Gateway	SIP	SIP IA Service Broker to Application Server	MSF-IA-SIP.006-FINAL (msf2003.063)

These protocols are shown in figure 4 below.

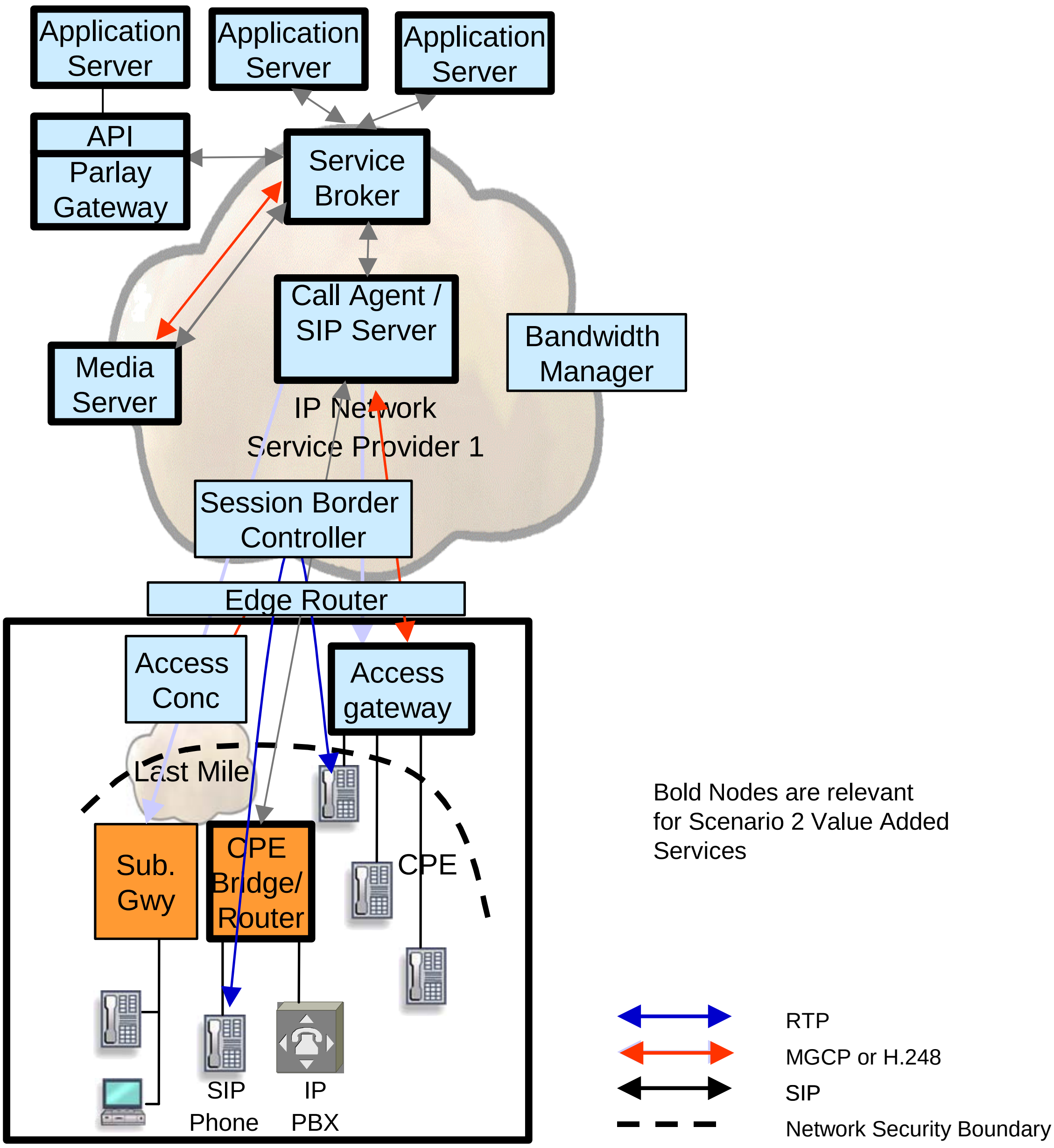

Figure 4: Scenario 2 Protocols

4.8 Redundancy and Failover Requirements

As for scenario 1 with the following additions

- Service Plane Failover – when Call Agent is not able to reach the primary Service Broker, failover to a back-up Service Broker should be enabled.
- Application Plane failover – Ability to reach a back-up app server if primary app server is unavailable
- Media Server Fail-over – Ability to reach a back-up media server if primary media server is unavailable

4.9 Feature List

Scenario 2 does not repeat the entirety of the feature testing included in scenario 1, instead it revalidates a small subset of the features available in scenario 1 and adds a set of Value Added Services using the Service Broker, Media Server and Application Servers.

The aim of the GMI 2004 is demonstrate how services can reside in multiple locations including Call Agents, SIP App Servers, and Parlay Gateways and Applications. The following feature list is kept relatively simple and includes recommendations as to what types of service should be provided from which elements.

- Call Agent: Basic Line side feature – revalidating Scenario 1 features
  - Call Waiting (Incoming call indicated using an in-band tone but no display of Calling Number or Name)
  - Call Transfer
  - Calling Number and Name delivery

- SIP Application Server: Network services
  - IP Conferencing
  - Originating and Terminating Screening Services
  - Number translation (i.e. VPN or FreePhone)
  - Voice Mail

- Parlay Gateways and Application Server: Network services using GCC and MPCC SCF's
  - Click-to-connect
  - Click- to-conference
  - GETS
    - ETS-enabled Click-to-connect
    - ETS-aware number translation

- Service Broker
  - No specific service (infrastructure supporting appropriate service identification and coordination)

- Media Server
  - Generic Conference Bridge
  - Announcement Servers
  - Voice Mail Media Processing

4.10 Exception Testing

The following exceptions tests are included in scenario 2.

- Priority /Emergency calling when MPLS LSP bandwidth exhausted

- Priority /Emergency calling when Call Agent loaded to capacity

4.11 Management - Subscriber Service Provisioning

A desirable objective is to demonstrate how a subscriber's service data is distributed to the appropriate Call Agent, Service Broker, and Application Servers via a single point flow-through provisioning mechanism.

5 Scenario 3 - Call Agent / SIP Server to PSTN within a single IP Network SP domain

Scenario 3 builds on scenario 2 to add PSTN interworking to scenario 2. Value Added Services are still included. This scenario also includes the treatment of priority calls.

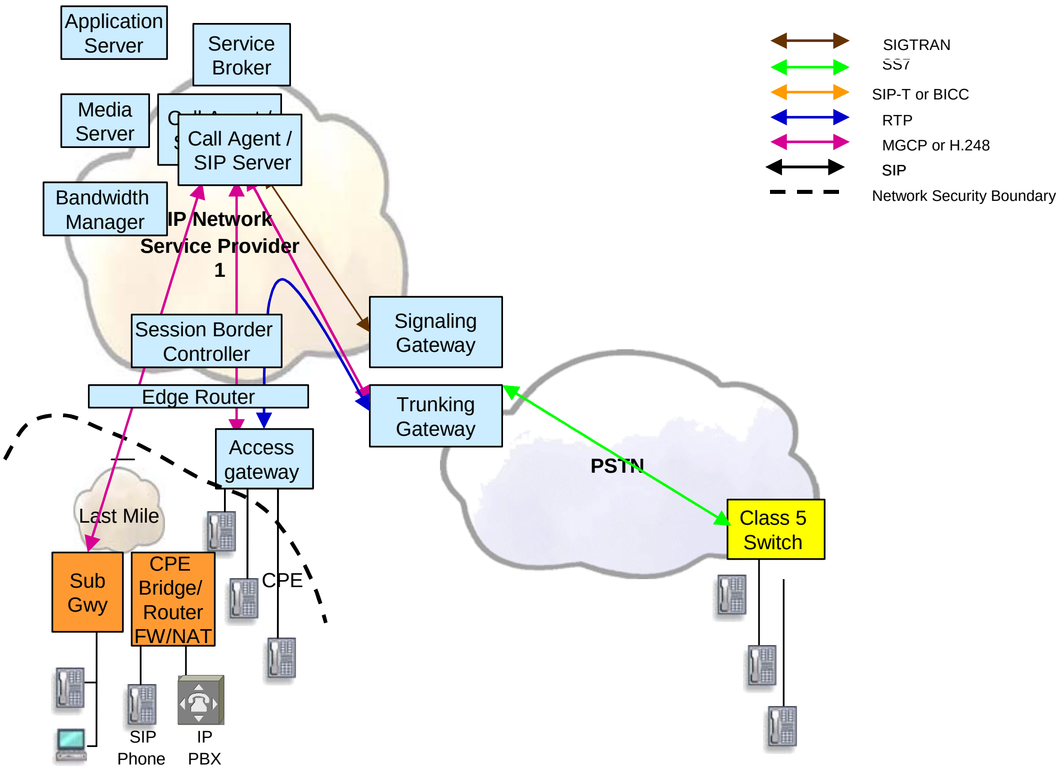

Figure 5: Scenario 3 Physical Architecture

The following sections add requirements and features to Scenarios 1 and 2. Each section notes where the requirements of scenarios 1 and 2 apply without change.

5.1 Domains

As for scenarios 1 and 2.

5.2 Network Components

In addition to the components listed in scenarios 1 and 2, the following components are added in scenario 3

- An optional additional Call Agent/SIP Server to control Signaling Gateway and Trunking Gateway. In this scenario the other Call Agent/SIP Server is used to control local customer equipment and access gateways.
  Alternatively a single Call Agent/SIP Server can manage both local customer equipment and the Signaling and Trunking Gateway.
- Signaling Gateway.
- Trunking Gateway.

In addition either a Class 5 switch (Digital Local Exchange DLE) or emulator is required for this scenario.

5.3 Network Links

5.3.1 Last Mile

As for scenarios 1 and 2.

5.3.2 Core IP Network

As for scenario 2 with the following addition.

Components such as the Signaling Gateway and Trunking Gateway expected to be connected via Ethernet, but can also use ATM, MPLS or other layer 2 protocol.

5.4 NAT / Firewall

As for scenarios 1 and 2.

5.5 Security

As for scenarios 1 and 2 with the following addition.

No security is required for Signaling Gateways or Trunking Gateways.

5.6 Quality of Service

As for scenarios 1 and 2.

In particular the following is included in scenario 3:

- In the core IP Network two or more MPLS LS Edge Routers controlled by a Bandwidth Manager using call priority, set by IAM/CPC and MTP/CPL in ISUP, and by the Resource Priority Header in SIP.

5.7 Protocols

As for scenarios 1 and 2 with the following additions.
Note: where the IAs are not yet published the current draft contribution is referenced in parentheses.

Interface	**Protocol**	**Implementation Agreement**	**MSF Contribution (Draft Contribution)**
Call Agent <-> Call Agent	SIP	SIP IA Call Agent to Call Agent	MSF-IA-SIP.004-FINAL (Not yet available)
	SIP-I (SIP-T)	Implementation Agreement for SIP-T Profile for Media Gateway Controller	MSF-IA-SIP-T.001.02-FINAL (msf2004.030)
Call Agent <-> Signaling Gateway	SIGTRAN (M3UA, M2UA or M2PA)	No MSF IA as this is expected to be an intra-vendor interface. See below for relevant specifications.	N/A
Call Agent <-> Trunking Gateway	H.248	Implementation Agreement for MEGACO/H.248 Profile for a Media Gateway Controller/Trunking Gateway using IP Trunks	MSF-IA-MEGACO.003-FINAL
	MGCP	Implementation Agreement for a MGCP Profile between a Call Agent and Trunking Gateway	MSF-IA-MGCP.003-FINAL (msf2004.029)
Class 5 switch <-> Signaling Gateway	SS7	Not required	N/A

NOTE: SIP is expected to support and exercise the Resource Priority Header. SS7 is expected to support and exercise IAM/CPC and MTP/CPL.

5.7.1 SS7 National Variants

The following national variants of SS7 signaling are used in each of the test locations:

- US Lab - ANSI (does not require equal access signaling)
- UK Lab - UK
- Japanese Lab - Japanese
- Korean Lab - ITU-T

5.7.2 Sigtran standards

The following standards are relevant for the Call Agent <-> Signaling Gateway

- SCTP
  - RFC 2960
    Stream Control Transmission Protocol
  - RFC 3309
    Stream Control Transmission Protocol (SCTP) Checksum Change
  - ETSI TS 102 144 V1.1.1 (2003-05)
    Services and Protocols for Advanced Networks (SPAN);MTP/SCCP/SSCOP and SIGTRAN (Transport of SS7 over IP);Stream Control Transmission Protocol (SCTP) [Endorsement of RFC 2960 and RFC 3309, modified]
- M2UA
  - RFC 3331
    Signaling System 7 (SS7) Message Transfer Part 2 (MTP2) - User Adaptation Layer
  - ETSI TS 102 141 V1.1.1 (2003-05)
    Services and Protocols for Advanced Networks (SPAN);MTP/SCCP/SSCOP and SIGTRAN (Transport of SS7 over IP);Message transfer part 2 User Adaptation layer (M2UA) [Endorsement of RFC 3331 (2002), modified]
- M2PA
  - draft-ietf-sigtran-m2pa-10.txt
    SS7 MTP2-User Peer-to-Peer Adaptation Layer
- M3UA
  - RFC 3332
    Signaling System 7 (SS7) Message Transfer Part 3 (MTP3) - User Adaptation Layer (M3UA)
  - ETSI TS 102 142 V1.1.1 (2003-05)
    Services and Protocols for Advanced Networks (SPAN);MTP/SCCP/SSCOP and SIGTRAN (Message of SS7 over IP);Message transfer part 3 User Adaptation layer (M3UA) [Endorsement of RFC 3332 (2002), modified]

5.8 Redundancy and Failover Requirements

As for scenarios 1 and 2.

5.9 Call Routing

Scenario 4 includes support for intra-domain call routing via SIP or SIP-I (SIP-T) encapsulating the relevant national SS7 variant.

5.10 Feature List

Scenario 3 repeats a substantial subset of the features tested in scenario 1 but making calls to and from the PSTN. It also revalidates a small subset of the features tested in scenario 2 as well as adding specific PSTN features.

- Basic Call with the following combinations
  - Access Gateway – Class 5 switch (black phone - black phone)
  - Access Gateway – Class 5 switch (fax - fax)
  - Access Gateway – Class 5 switch (V.90 modem - V.90 modem)
  - IP PBX – Class 5 switch (black phone)
  - SIP Phone – Class 5 switch (black phone)
  - Subscriber Gateway - Class 5 switch (black phone - black phone)
  - Subscriber Gateway - Class 5 switch (fax - fax)
  - Subscriber Gateway - Class 5 switch (V.90 modem - V.90 modem)

- Support for Residential Features calling to and from the PSTN
  - Call Transfer
  - Call Waiting (Incoming call indicated using an in-band tone but no display of Calling Number or Name)
  - Call Diversion (Call Forwarding)
  - Calling Number and Name Delivery
  - Three-party conference

- Support for Business Features
  - Centrex
  - Private numbering plan
  - DDI / DDO

- SIP Application Server: Network services
  - Number translation (i.e. VPN or FreePhone) for PSTN number

- Parlay Gateways and Application Server: Network services using GCC and MPCC SCF's
  - Click-to-connect to PSTN phone

- GETS
  - Access Gateway black phone to PSTN black phone via IP domain 1
  - Caller authentication in the IP domain
  - Call Admission with priority
  - Priority mapping and interworking between SS7 and SIP

- Emergency
  - SIP Phone to PSTN operator

5.11 Exception Testing

Not included in scenario 3.

6 Scenario 4 - Call Agent / SIP Server across multiple IP Network Service Provider domains and to PSTN

Scenario 4 adds VoIP interworking between IP Network Service Provider domains. It builds on Scenario 3 but removes the Value Added Services and the Priority Services from scenario 3 from scenario 2. This scenario is shown in figure 6 below.

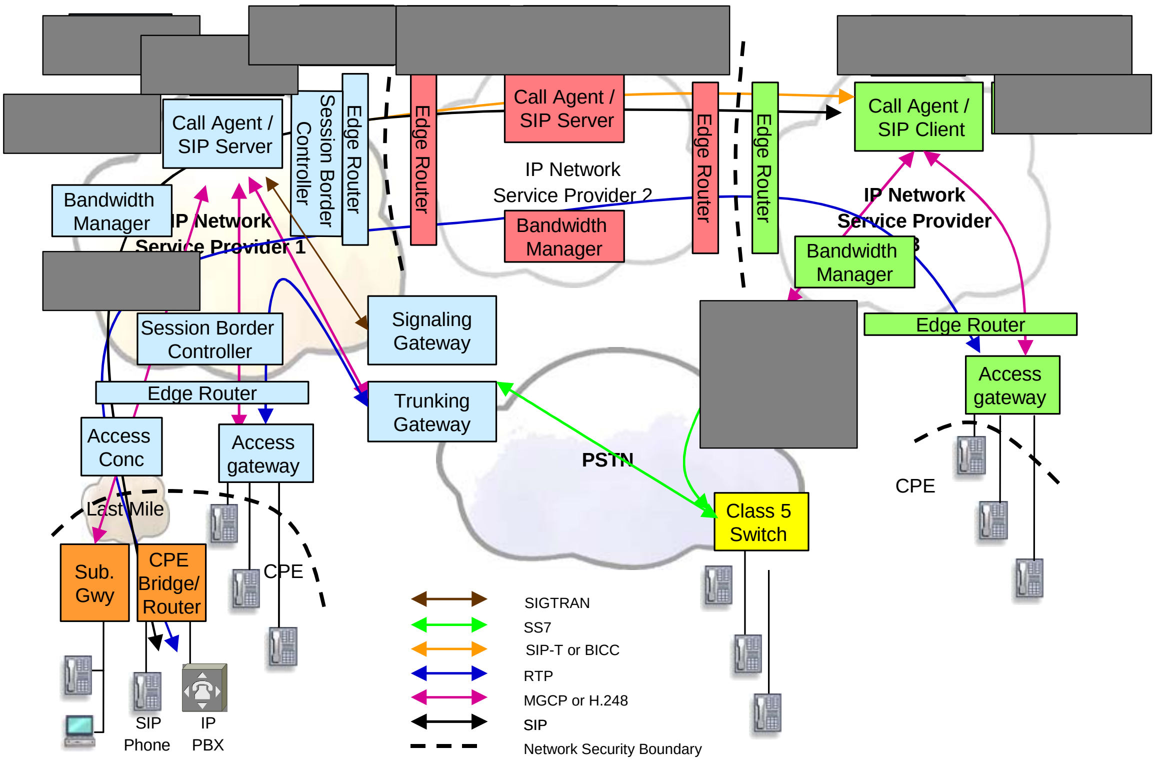

Figure 6: Scenario 4 Physical Architecture

Note: For simplicity this diagram omits Session Border Controller's from IP Network Service Provider 2 and IP Network Service Provider 3.

6.1 Domains

Scenario 4 includes three IP networks connecting three service provider domains.

- IP Network 1 is managed by Service Provider 1 and provides local access.
- IP Network 2 is managed by Service Provider 2 and is providing inter-exchange carrier (IXC) access.
- IP Network 3 is managed by Service Provider 3 is offering local access.

Each IP network can either run in a public or private IP address space.

6.2 Network Components

This scenario adds the following components to scenario 3

- Additional Call Agent/SIP Server to control local customer equipment and access gateways in IP Network 3.
- Additional Call Agent/SIP Server to control traffic transiting through IP Network 2.
- An additional Bandwidth Manager in both IP Networks 2 and 3 to provide end-to-end QoS.

- An additional Session Border Controller in both IP Networks 2 and 3 to support NAT between domains.
- Additional Edge Routers in both IP Networks 2 and 3.
- Additional access device(s) in IP Network 3.

Some of the Call Agents/SIP Servers can be combined, for example a single Call Agent could provide control of both the local customer equipment and the Signaling Gateway/Trunking gateway in IP Network 1.

In addition this scenario removes the components added in scenario 2. These are

- Application Servers.
- Parlay Gateways and Application Servers.
- Service Brokers.
- Media Servers.

6.3 Network Links

6.3.1 Last Mile Networks

As for scenario 3 in both IP Networks 1 and 3.

6.3.2 Core IP Networks

As for scenario 3 in all 3 IP Networks.

6.3.3 Inter-Domain Network

Between the IP Networks (i.e between IP Network 1 and IP Network 2 and between IP Network 2 and IP Network 3), the links will be provisioned IP links of fixed known bandwidth. Any layer 2 including INTERNET 2 can be used, however MPLS is not expected to be available for these inter-domain links.

6.4 NAT / Firewall

In addition the NAT and Firewall support in scenario 1, scenario 4 also includes Network Address Translation (NAT) and Firewall function between the 3 IP Networks. In particular each IP Network can run in its own private IP address space and use NAT for inter-domain connectivity.

This scenario uses either of the following solutions to cope with NAT and Firewall at both the customer premise and inter-domain.

- A Session Border Controller which sits on both the media and signaling path.
  For SIP(-T) the Session Border Controller acts as a B2BUA and both translates the IP addresses in the SIP(-T) signaling and redirects the RTP traffic.
- A firewall/RTP proxy device controlled via the Pinhole Control Protocol described in the Implementation Agreement for RTP Proxy/FW Control Protocol (msf2003.113.00)

6.5 Security

As for scenario 1, 2 and 3 with the following additions.

Scenario 4 also includes the following security solutions for SIP(-T) between IP networks.

- HTTP Digest for authentication
- TLS for encryption and Integrity
- S/MIME encryption is optional

6.6 Quality of Service

As for scenario 1 with the following addition.

- The inter-domain links are of a fixed known bandwidth and do not utilize MPLS. The Bandwidth Managers must perform Call Admission Control based on the link bandwidth and number of calls already in progress.

6.7 Protocols

As for scenario 3, except that the protocols specified in scenario 2 are not required and the SIP Resource Priority Header is not required.

6.7.1 SS7 National Variants

The following national variants of SS7 signaling are used in each of the test locations:

- US Lab - ANSI (does not require equal access signaling)
- UK Lab - UK
- Japanese Lab - Japanese
- Korean Lab - ITU-T

For inter-domain links between test locations the ITU-T variant of SS7 signaling is used.

6.8 Call Routing

Scenario 4 includes support for multi-domain call routing via SIP or SIP-I (SIP-T) encapsulating ITU-T ISUP

6.9 Redundancy and Failover Requirements

As for scenarios 1, 2 and 3.

6.10 Feature List

Scenario 4 repeats a substantial subset of the features tested in scenario 1 but making calls between IP networks. It also revalidates a small subset of the features tested in scenario 3 for calls to and from the PSTN.

The tests are run in the following multi-domain setups.

- Case 1: Originating calls in IP Network 1 and terminating calls in IP Network 3, transiting IP Network 2

- Case 2: Originating calls in IP Network 3 and terminating calls in IP Network 1, transiting IP Network 2

The following features are included in scenario 4 for multi-domain IP <-> IP tests.

- Basic Call with the following combinations
  - Access Gateway - Access Gateway (black phone - black phone)
  - Access Gateway - Access Gateway (fax - fax)
  - Access Gateway - Access Gateway (V.90 modem - V.90 modem)
  - IP PBX - Access Gateway (black phone)
  - SIP Phone – Access Gateway (black phone)
  - SIP Phone – SIP Phone
  - Subscriber Gateway - Access Gateway (black phone - black phone)
  - Subscriber Gateway - Access Gateway (fax - fax)
  - Subscriber Gateway - Access Gateway (V.90 modem - V.90 modem)
  - Subscriber Gateway (black phone) – SIP Phone
  - IP PBX – SIP phone
  - IP PBX - Subscriber Gateway (black phone)
  - IP PBX – IP PBX
  - Subscriber Gateway - Subscriber Gateway (black phone - black phone)
  - Subscriber Gateway - Subscriber Gateway (fax - fax)

  - Subscriber Gateway - Subscriber Gateway (V.90 modem - V.90 modem)

- Support for Residential Features
  - Call Transfer
  - Call Waiting (Incoming call indicated using an in-band tone but no display of Calling Number or Name)
  - Call Diversion (Call Forwarding)
  - Calling Number and Name Delivery
  - Three-party conference

- Support for Business Features
  - Centrex
  - Private numbering plan
  - DDI / DDO

The following features are included in scenario 4 for multi-domain IP <-> PSTN tests. These test originate calls in IP Network 3, traverse IP Networks 2 and 1 and then are routed to the PSTN.

- Basic Call with the following combinations
  - Access Gateway – Class 5 switch (black phone - black phone)
  - Access Gateway – Class 5 switch (fax - fax)
  - Access Gateway – Class 5 switch (V.90 modem - V.90 modem)

- Support for Business Features
  - Centrex
  - Private numbering plan
  - DDI / DDO

- GETS
  - Access Gateway black phone to PSTN black phone
  - SIP Phone to PSTN black phone, with PIN collection and authentication in the PSTN domain

- Emergency
  - SIP Phone to PSTN operator
  - Access Gateway black phone to PSTN black phone
    - Continuous re-try
    - Emergency call report / alarm
    - Call hold (clearing program modification)

6.11 Exception Testing

The following exceptions tests are included in scenario 4.

- Rejection of Calls when inter-domain links are bandwidth exhausted

7 Scenario 5 - Call Agent / SIP Server across multiple IP Network Service Provider domains and to PSTN with Value Added Services

This scenario adds Value Added Services to scenario 4 using Service Brokers, Media Servers, Application Servers and Parlay Servers. This is shown in figure 7 below.

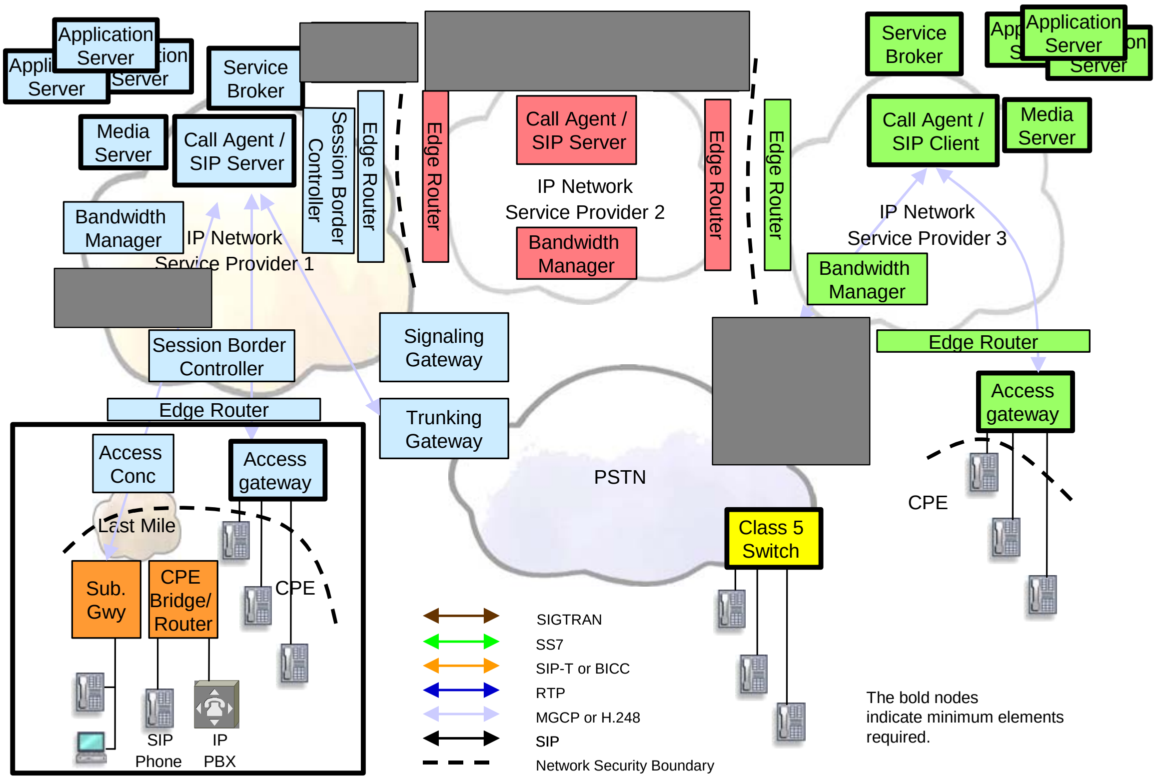

Figure 7: Scenario 5 Physical Architecture

Note: For simplicity this diagram omits Session Border Controller's from IP Network Service Provider 2 and IP Network Service Provider 3.

The following sections add requirements and features to Scenarios 1, 2, 3 and 4. Each section notes where the requirements of scenarios 1, 2, 3 and 4 apply without change.

7.1 Domains

As for scenario 4.

7.2 Network Components

In addition to the components listed in scenario 4 the following components are added in scenario 5

- One or more redundant Application Servers in IP network 1 in service provider domain 1.
- One or more redundant Application Servers in IP network 3 in service provider domain 3.
- One or more redundant Parlay Gateways and Application Servers in IP network 1 in service provider domain 1.
- One or more redundant Parlay Gateways and Application Servers in IP network 3 in service provider domain 3.

- One or more redundant Service Brokers in IP network 1 in service provider domain 1.
- One or more redundant Service Brokers in IP network 3 in service provider domain 3.
- One or more redundant Media Servers in IP network 1 in service provider domain 1.
- One or more redundant Media Servers in IP network 3 in service provider domain 3.

Some of these network components can be combined, for example Media Server and Application Server or Service Broker and Call Agent.

For the remainder of this section the term Application Servers includes Parlay Gateways.

7.3 Network Links

7.3.1 Last Mile Networks

As for scenario 4.

7.3.2 Core IP Networks

As for scenario 4 with the following addition.

Components such as the Service Broker, Application Server and Media Server are expected to be connected via Ethernet, but can also use ATM, MPLS or other layer 2 protocol.

7.3.3 Inter-Domain Network

As for scenario 4.

7.4 NAT / Firewall

As for scenario 4.

7.5 Security

As for scenario 4 with the following addition.

No security is required for Service Brokers, Application Servers or Media Servers.

7.6 Quality of Service

As for scenario 4, with the addition of the following:

- In the core IP Network two or more MPLS LS Edge Routers controlled by a Bandwidth Manager using call priority, set by IAM/CPC and MTP/CPL in ISUP, and by the Resource Priority Header in SIP.

7.7 Protocols

As for scenario 4 with the following additions.
Note: where the IAs are not yet published the current draft contribution is referenced in parentheses.

Interface	**Protocol**	**Implementation Agreement**	**MSF Contribution (Draft contribution)**
Call Agent <-> Service Broker	SIP	SIP IA Call Agent to Service Broker	MSF-IA-SIP.005-FINAL (msf2003.064)
Service Broker <-> Application Server	SIP	SIP IA Service Broker to Application Server	MSF-IA-SIP.006-FINAL (msf2003.063)
Service Broker <-> Media Server	SIP	SIP Media Server IA	MSF-IA-SIP.009-FINAL (msf2004.006)

Service Broker <-> Parlay Gateway	SIP	SIP IA Service Broker to Application Server	MSF-IA-SIP.006-FINAL (msf2003.063)
Service Broker <-> Service Broker	SIP	SIP IA Service Broker to Service Broker	MSF-IA-SIP.007-FINAL (msf2003.065)

NOTE: SIP is expected to support and exercise the Resource Priority Header. SS7 is expected to support and exercise IAM/CPC and MTP/CPL. These protocols are shown in figure 8 below.

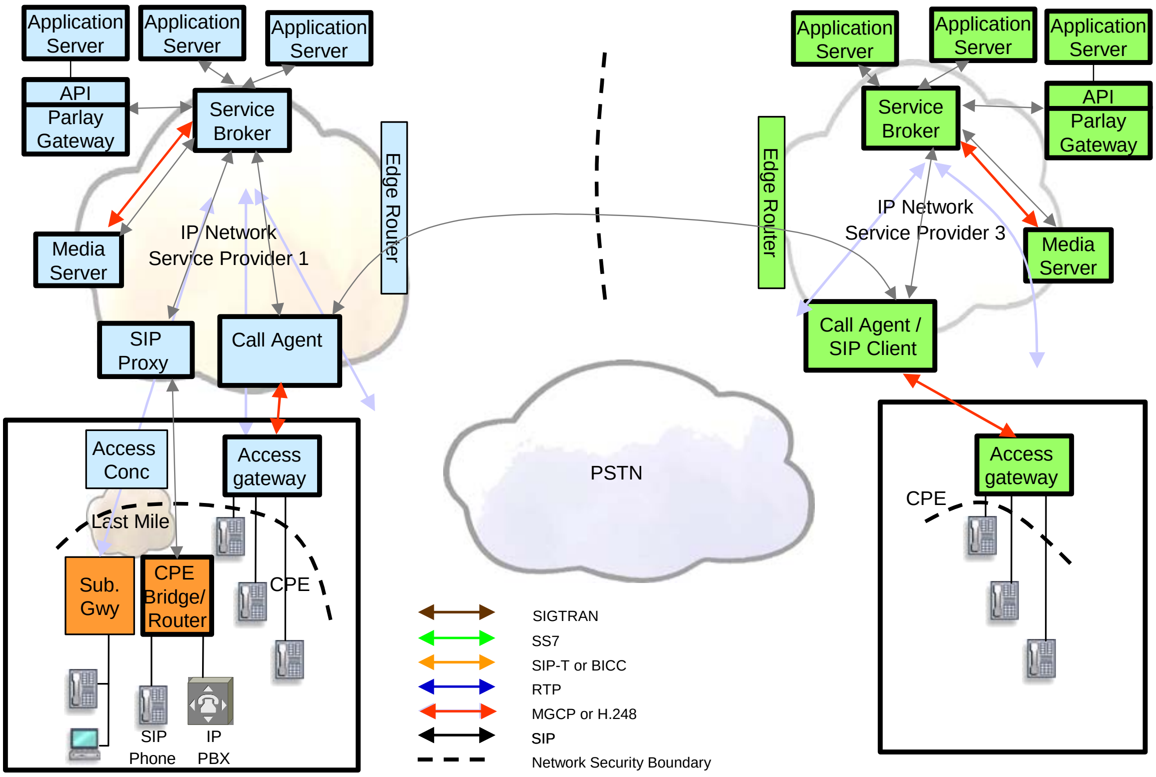

Figure 8: Scenario 5 Protocols

7.8 Call Routing

Scenario 5 includes support for multi-domain call routing via SIP or SIP-I (SIP-T).

7.9 Redundancy and Failover Requirements

As for scenario 4 with the following additions

- Service Plane Failover – when Call Agent is not able to reach the primary Service Broker, failover to a back-up Service Broker should be enabled.
- Application Plane failover – Ability to reach a back-up app server if primary app server is unavailable
- Media Server Fail-over – Ability to reach a back-up media server if primary media server is unavailable

7.10 Feature List

Scenario 5 repeats the feature testing performed in scenario 2 in a multi-domain environment.

The tests are run in the following multi-domain setups.

- Case 1: Originating Services in IP Network 1 and Terminating Services from IP Network 3

- Case 2: Originating Services from IP Network 3 and Terminating Services from IP Network 1

- Case 3: Mixed originating services from both IP Network 1 and IP Network 3 and Terminating services from both IP Network 1 and IP Network 3.

The following features are included in scenario 5.

- Call Agent: Basic Line side feature – revalidating Scenario 4 features across multiple domains.
  - Call Waiting (Incoming call indicated using an in-band tone but no display of Calling Number or Name)
  - Call Transfer
  - Calling Number and Name delivery

- SIP Application Server: Network services
  - IP Conferencing
  - Originating and Terminating Screening Services
  - Number translation (i.e. VPN or FreePhone)
  - Voice Mail

- Parlay Gateway and Application Server: Network services using GCC and MPCC SCF's
  - Click-to-connect
  - Click- to-conference
  - GETS
    - ETS-enabled Click-to-connect
    - ETS-aware number translation

- ETS/GETS
  - Access Gateway black phone to PSTN black phone via IP domains 1 and 2
  - Caller authentication in the IP domain
  - Call Admission with priority
  - Priority mapping and interworking between SS7 and SIP
  - Priority interworking between IP domains 1 and 2

- Service Broker
  - No specific service (infrastructure supporting appropriate service identification and coordination)

- Media Server
  - Generic Conference Bridge
  - Announcement Servers
  - Voice Mail Media Processing

7.11 Exception Testing

Not included in scenario 5.

7.12 Management - Subscriber Service Provisioning

A desirable objective is to demonstrate how a subscriber's service data is distributed to the appropriate Call Agent, Service Broker, and Application Servers via a single point flow-through provisioning mechanism.

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