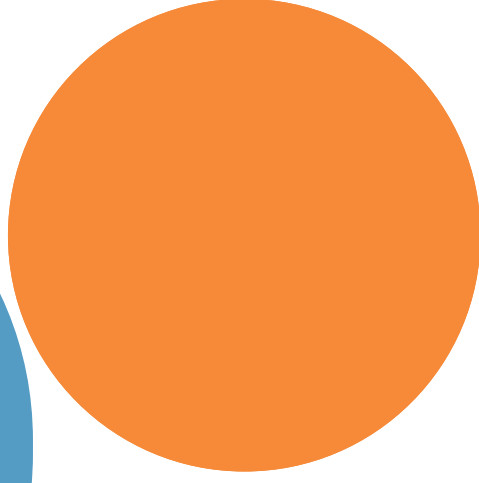
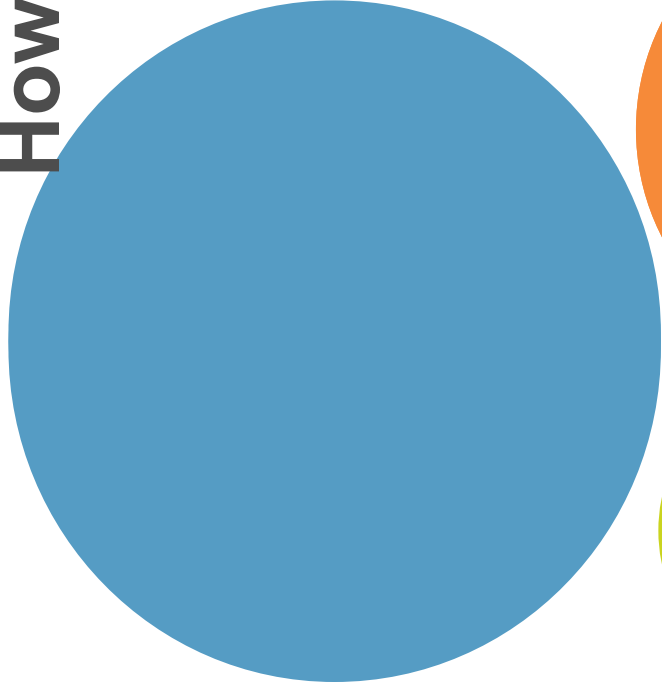


How to implement international IP Voice Interconnections different solutions for different needs



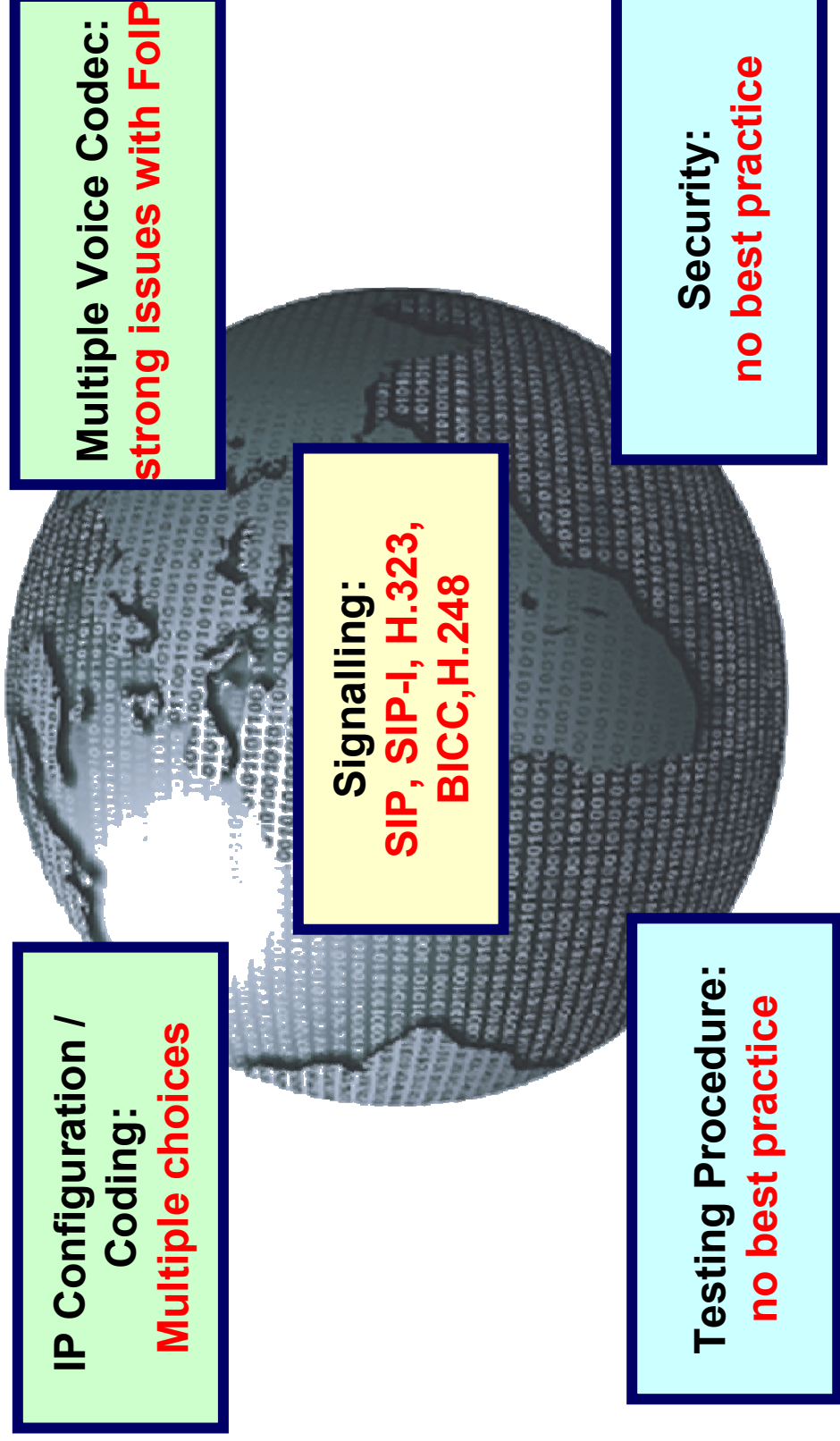
presented by
Alessandro Forcina
(i3 Forum WS Technical Aspects Chairman)
TELECOM ITALIA SPARKLE

May 26th, 2011

Agenda

- i3f guidelines
- The Business Framework
- Implementing and Testing an IP Interconnection
 - architecture
 - signalling protocols
 - media coding
 - QoS control
 - Testing procedures
- Voice over IPX
- Routing and Addressing

The Voice over IP world in 2007/08



**Too many alternatives, no clear path to be pursued,
no best practice**

Three guidelines for i3 forum

- Be as “*simple*” as possible
- Cover (the market) as much as possible
- Give examples / best practices



• WS “Technical Aspects” Mission and Objective

“To technically allow a worldwide and unrestrained migration to IP of existing TDM International voice interconnections **selecting**, on the basis of existing standards/recommendations issued by international bodies, a **unique network architecture capable to support one (or a limited number of) interconnection model(s) for the implementation of trusted, secure and**

QoS compliant VoIP interconnection between Int. Wholesale Carriers”

The Business Framework

Different Business Models for International Voice Services

- Bilateral
- Hubbing (Open Wholesale Market)
- IPX
- Others (e.g. Communities with on-net / off-net traffic concepts)

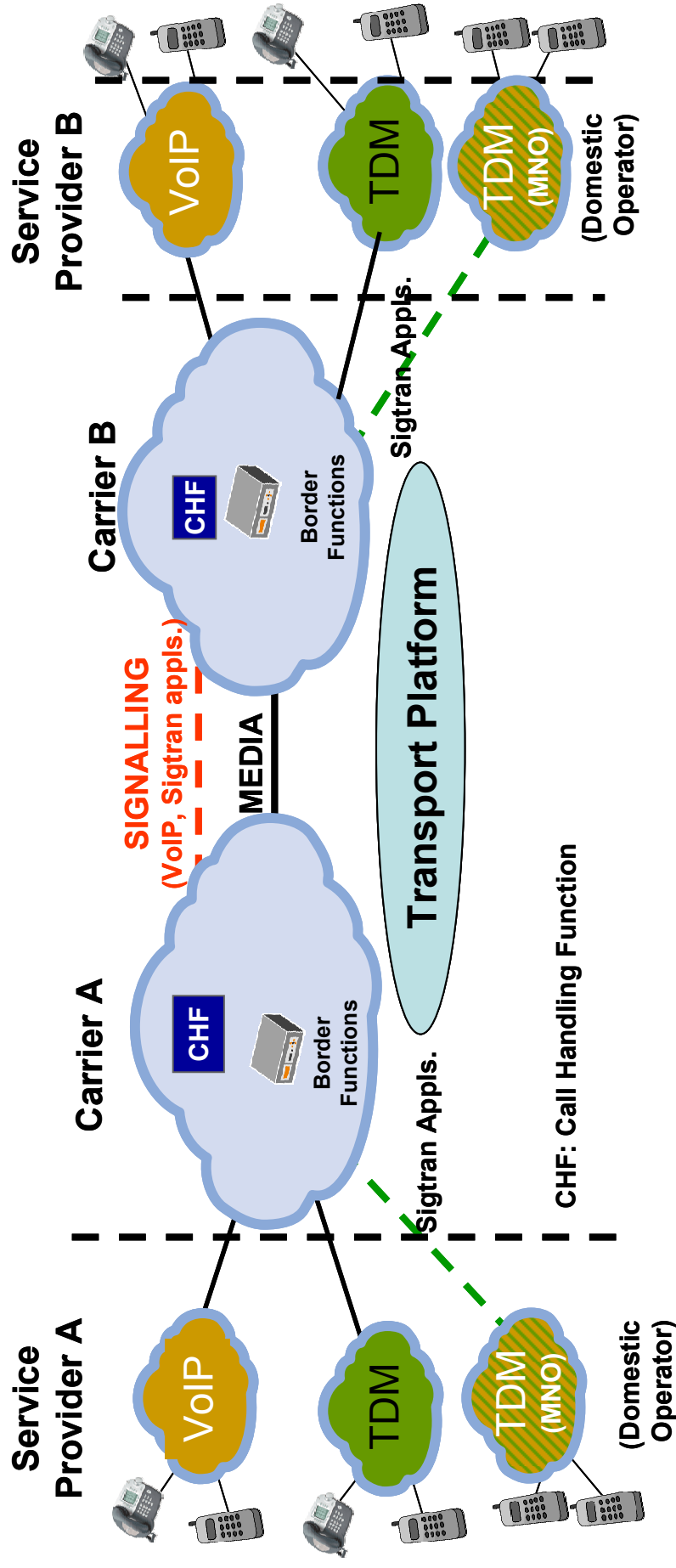
With

- Various level of quality (at the IP and service layers)
- Cascade payment
- Volume based billing (with or without commitments)



No major change between TDM business model vs. Voice over IP

General Reference Configuration

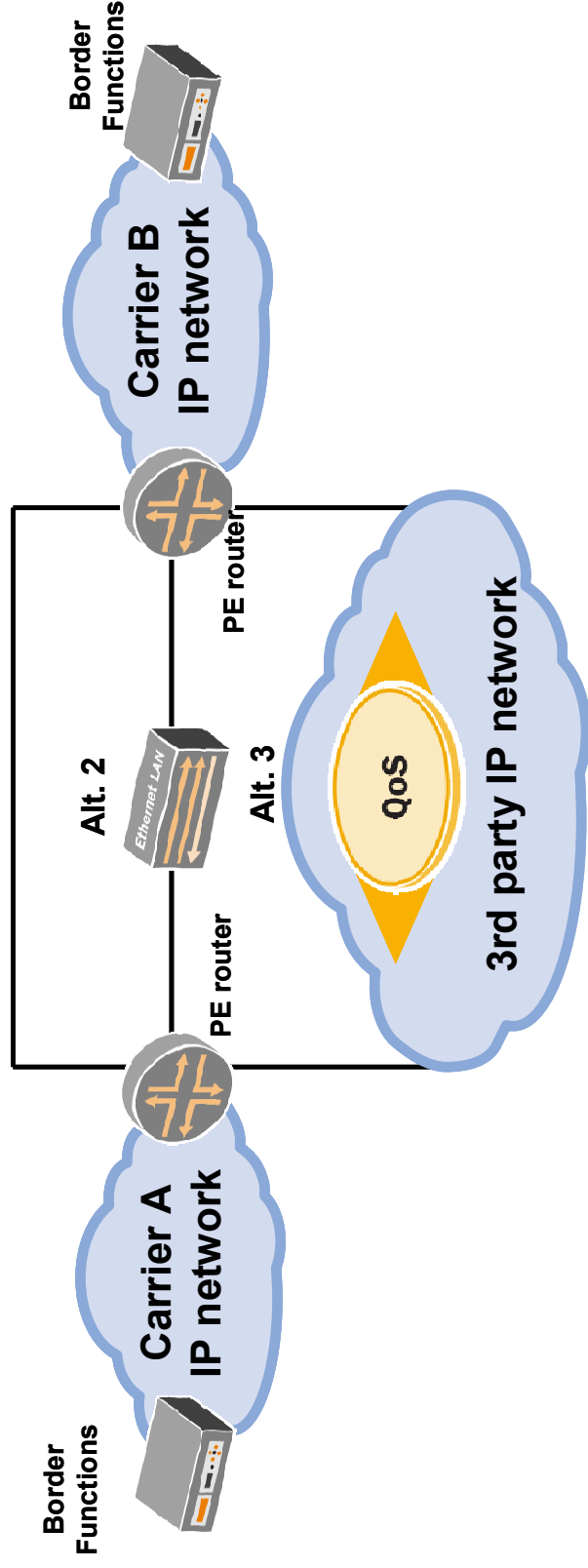


Domestic interconnections (TDM/IP; IP/IP) are out of scope

Which kind of interconnections? Private

• Private-oriented interconnection

- no unidentified third party is able to affect the VoIP service
- Transport Alternatives at Layer 1 / Layer 2 / Layer 3
Alt. 1



Private interconnections can replace existing TDM-based ones guaranteeing the highest level of quality both in terms of voice call quality, service quality, network availability and network security.

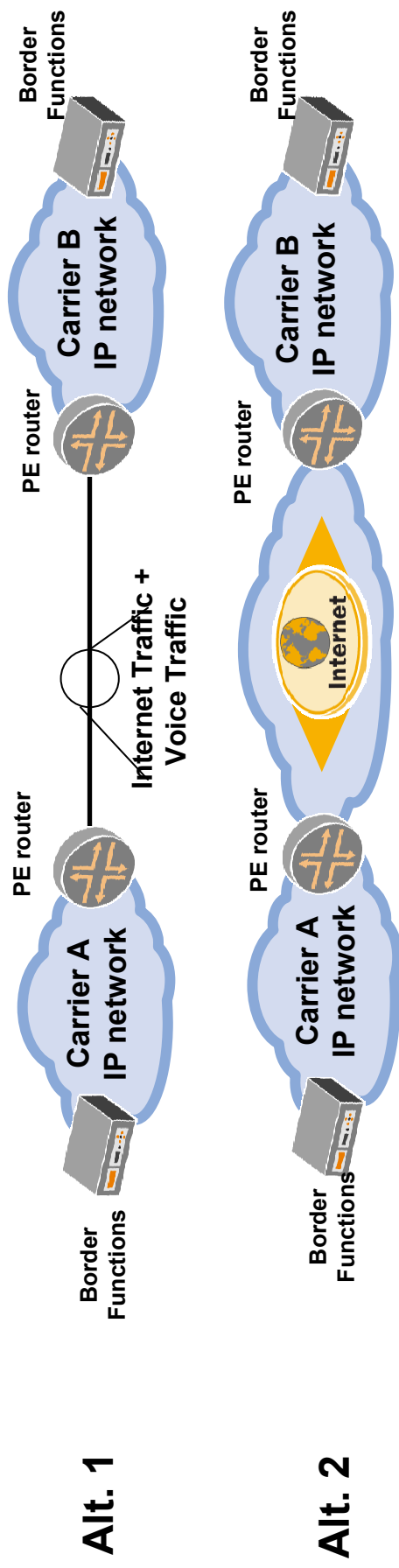
..as well as Public Interconnections

Public-oriented interconnection

VoIP traffic is mixed with other IP traffic coming from the Public Internet → gateways' interfaces can be reached from unidentified third parties. **VPN IP Sec encryption required for signalling**

Transport Alternatives:

- Alt. 1: direct interconnection sharing data and VoIP traffic
- Alt. 2: via Public Internet: non direct interconnection



Public interconnections imply lower cost and, in general, lower provisioning time

Which Signalling Protocol?

- **Signalling Protocols for Voice services: 2 protocols selected**
 - **SIP protocol** (IETF RFC 3261): a specific profile has been defined
 - **SIP-I protocol** ISUP enabled SIP profile (ITU-T Q.1912.5 Annex C Profile C), selected also for the support of ISDN services
- **Support of Sigtran for Mobile appl.s (SMS, Camel and roaming mobile)**
 - via **M2PA (preferred solution)** or M3UA when no relaying capability is needed
 - SCTP between IP layer and SIGTRAN adaptation layers
- **Mapping of ISUP RC \leftrightarrow SIP, SIP-I**
 - Lack of convergence among ITU Q.1912.5, IETF RFC3398, 3GPP TS 29.163
 - Impact on routing and QoS (NER) control
 - Adoption of **Reason Header** as temporary and partial fix (in 2010)
- **Converging on a unique ISUP RC \leftrightarrow SIP, SIP-I mapping**
 - Joint activity with 3GPP CT3 \rightarrow **approval in 3-11 of 3GPP TS 29.163 V.7.22**

Note: SCTP: Stream Control Transmission Protocol

Which coding scheme?

	MANDATORY	OPTIONAL
Narrow band Codecs	<ul style="list-style-type: none"> •ITU-T G.711 A-law, μ-law 64kbit/s •ITU-T G.729, G.729a, G.729b, G.729ab 8kbit/s 	<ul style="list-style-type: none"> •ITU-T G.723.1 •ITU-T G.726 •AMR-NB
Wide band Codecs	<ul style="list-style-type: none"> •ITU-T G.722 (for fixed operators) •AMR-WB (for mobile operators) 	
Low Bit Rate Codecs	<ul style="list-style-type: none"> •ITU-T G.729a with VAD/DTX 	<ul style="list-style-type: none"> •AMR-NB with VAD/DTX
Fax Modem Codecs	<ul style="list-style-type: none"> •FAX: ITU-T G.711 and ITU-T T.38 •Modem: ITU-T G.711 	

- Voice Engineering document Rel. 3 (Q2 '11)
- Codec selection confirmed by internal i3 forum survey (Q4 '10)
- White paper on Fax over IP in co-operation with Sip Forum (Q1 '11)
- Fax over IP testing activity in progress (Q2 '11) → results in FoIP Rel. 2 (Q4 '11)

Note: Using G.723.1 quality impairments have to be considered

...but the interconnection must be secure

- **Border Functions (e.g. Session Border Controller)**
 - **It is strongly recommended that Border Functions be always implemented,** achieving topology hiding and NAT/NAPT translation

Security Threats	DoS/DDoS Attack, Protocol Vulnerabilities, Address/Identity Spoofing, Theft of Service, Rogue Media, Session Hijacking, Network Intrusion, Internal Network Security
Security Mechanisms	Topology Hiding, Encryption, Authentication, Access Control Lists, Reverse Path Filters, Traffic policing, Application Level Relaying, Deep Packet Inspection, SRTP, DNSSEC, Media Filtering, Firewalls, Intrusion Detection Systems, Device Hardening, Logging and Auditing, Security Information & Code Updates

- **White paper on Security for IP Interconnections delivered at ITW2011(Q2 '11)** with Recommendation Matrixes
 - **3 level of security: Basic, i3f Recommended, i3f Optional**
 - **covering signalling, query and media interfaces**

... and with controlled QoS

BASIC REQUIREMENT

(**ALL** claim about Quality of Service Control)



NO SOLUTION YET

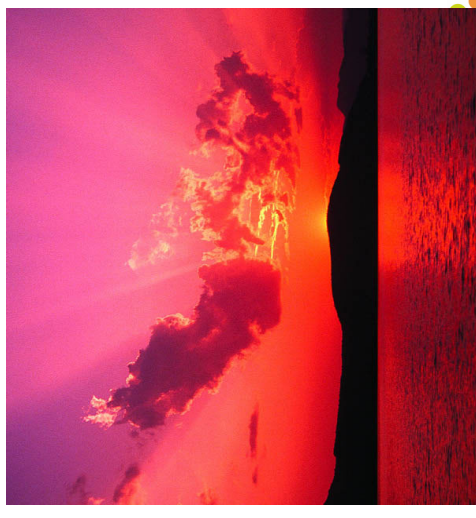
(at least for **ALL** Business Model, e.g. Bilateral, IPX, Open WS Market,
and on large scale)

SPs (IPX SPs) kill Carriers (IPX Providers)




Cyfemestra kills Cassandra

A new dawn



QoS: what has been defined

Scope

-  Monitoring (supervision) against given thresholds
-  Troubleshooting
-  Service Level Agreement (SLA) and Quality of Service reporting

Parameters Identification and relevant definition

Transmission/IP parameters:

- ✓ RTP round-trip delay
- ✓ RTP jitter
- ✓ RTP packet loss

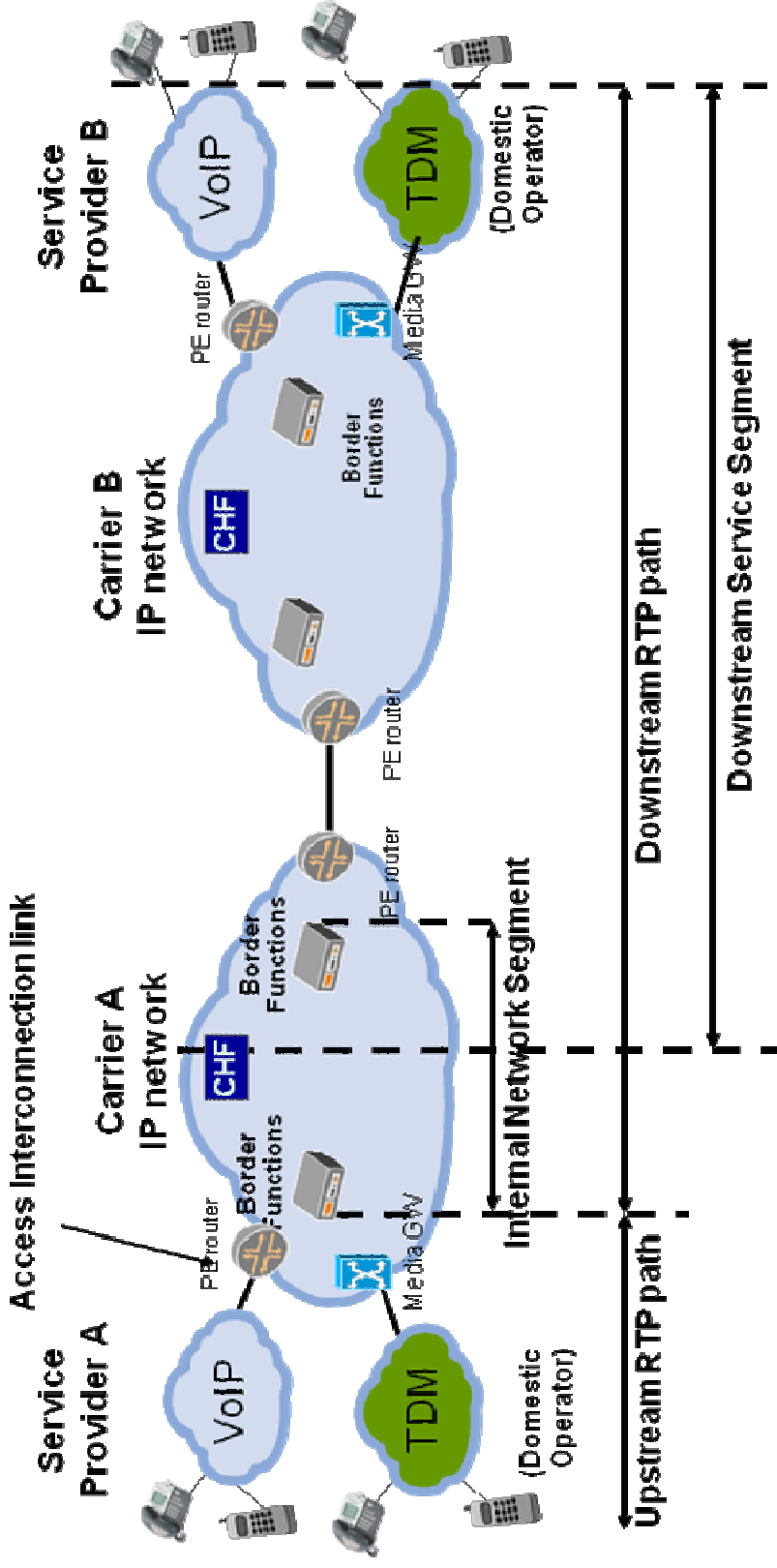
Service parameters

- ✓ MOS_{CQE} / R-Factor
- ✓ ALOC
- ✓ ASR
- ✓ NER
- ✓ PGRD

-  CLI transparency not considered a KPI however, it is strongly recommended that international carriers will pass on CLI unaltered.

QoS: what has been proposed

Reference Configuration for QoS for Service Provider <-> Carrier relationship



CHF: Call Handling Functions

Note: it is possible that more than two Carriers can be involved in the Service Provider-to-Service Provider communication. If more than two Carriers are involved, Carrier B is meant to be the last in the path, i.e. the Carrier interconnecting to Service Provider B. Consequently, Carrier A and Carrier B may not have a direct relationship.

A corresponding configuration is given for the Carrier-to-Carrier relationship

QoS: Various Alternatives

Quality Measurements Aggregation

-  adding network by network the measured quality data
- i3f WS Service analysed the commercial / operational impact of such a scheme

RTPCP-Based

-  existing i3 Forum Interconnect Model (proposed in 2010)

Alternative Loop-back

-  the Border Functions (i.e. SBC) makes testing calls to dummy numbers terminated to the last SBC, based on DRAFT IETF -mmusic-loopback-14

SIP-based *quality feedback*

-  the SIP message (INFO or PUBLISH) send back the quality information each carrier's intermediate and final (or far-end) border functions would append and return transport and service parameters

Making Testing more efficient

- Need to have
 - **clear guideline** → saving \$\$\$ / €€€ in provisioning time
 - **skilled staff** → saving \$\$\$ / €€€ in training
 - **smaller staff** → saving \$\$\$ / €€€ in salaries
 - **short troubleshooting** → saving \$\$\$ / €€€ in network operation

From i3f WS “Operations”

• *Migration Interconnection Form*

A powerful tool to exchange technical information for the set up of a new VoIP interconnection and understand at first glance possible compatibility issues

• *Interoperability Test Plan*

A detailed test list supporting testing phase during the activation of a new interconnection or during troubleshooting phase

• *Training Program in operation / Best practices guide (to be released)*

Handbook with operational guidelines from reference configurations to provisioning to testing

... and monitoring the migration process

• Monitoring the development of the IP migration (i3f surveys)

- Latest survey shows a **range between 20% and 40% of VoIP traffic** vs. total voice traffic
- Incoming VoIP traffic is generally slightly higher than terminated VoIP traffic

• Tracking of ongoing issues → highlighting possible interworking issues between vendors

• Case Study on adoption of

- SIP vs SIP-I
- Private Interconnection vs Public Interconnection

Voice over IPX

• Need to have

open, multiservice, secure, QoS controlled, cascading payments IP platform

IP eXchange from GSMA

Considering the market (carriers' platforms and customers' requirement)

• Technical Specification for Voice over IPX services (Oct. 2010)

- focus on the Multilateral Hubbing connectivity mode
- provides specifications which, meeting the basic GSMA requirements, can be implemented for IP routing, signalling, media, security, QoS control and service routing
- differentiates from current GSMA specification on specific topics (e.g. break-in/out)

• Understanding the different flavours of IP eXchange (May 2011)

- presents the differences of the main various forms of IPX networks and solutions that are being offered in the market so far

Routing & Addressing

 Need to have

NP Resolution (F+M) to exploit advanced routing/addressing schemes

Efficient, cost effective systems / service providers

Multiple solutions already available in the market

Techniques for Carriers' Advanced Routing and Addressing Schemes

 technical requirements for the query, provisioning and replication interfaces

- **ENUM/DNS, SIP Redirect**
- SS7 MAP/TCAP, DIAMETER

 requirements for Service Provider Identity (SPID)

IP Routing Directory basic requirements

- type of information to be stored
- security and accounting
- data partitioning
- scalability
- QoS requirements

Routing & Addressing: a standardization issue

 **Service Provider Identity (SPID) needs worldwide standardization, today proprietary solution available in the market**

Basic SPID requirements (from i3f WS “Service Requirements”)

- Globally unique.
- Flat in structure.
- Only numeric digits (0-9).
- Fixed length.
- At least 8 numeric digits (giving 100 million possible identifiers)

Administratively (a subset is given)

- Global SPID identifiers should be assigned by an international assignment body
- Entities should be able to apply for multiple Global SPID identifiers
- For carriers that wish to define a SPID based on their ITU-T E.212 assignment, consistency has to be ensured

ITU-T SG2, IETF, IANA involved in the standardization process

i3 Forum urges the finalisation of this process and is ready to support it

The Voice over IP world in 2011

**Voice over IPX
Technical Specs.
Analysis market offers**

**Routing & Addressing
White paper in 2010
SPID Requirements**

**IP Configuration /
Coding:
Private / Public Conf.**

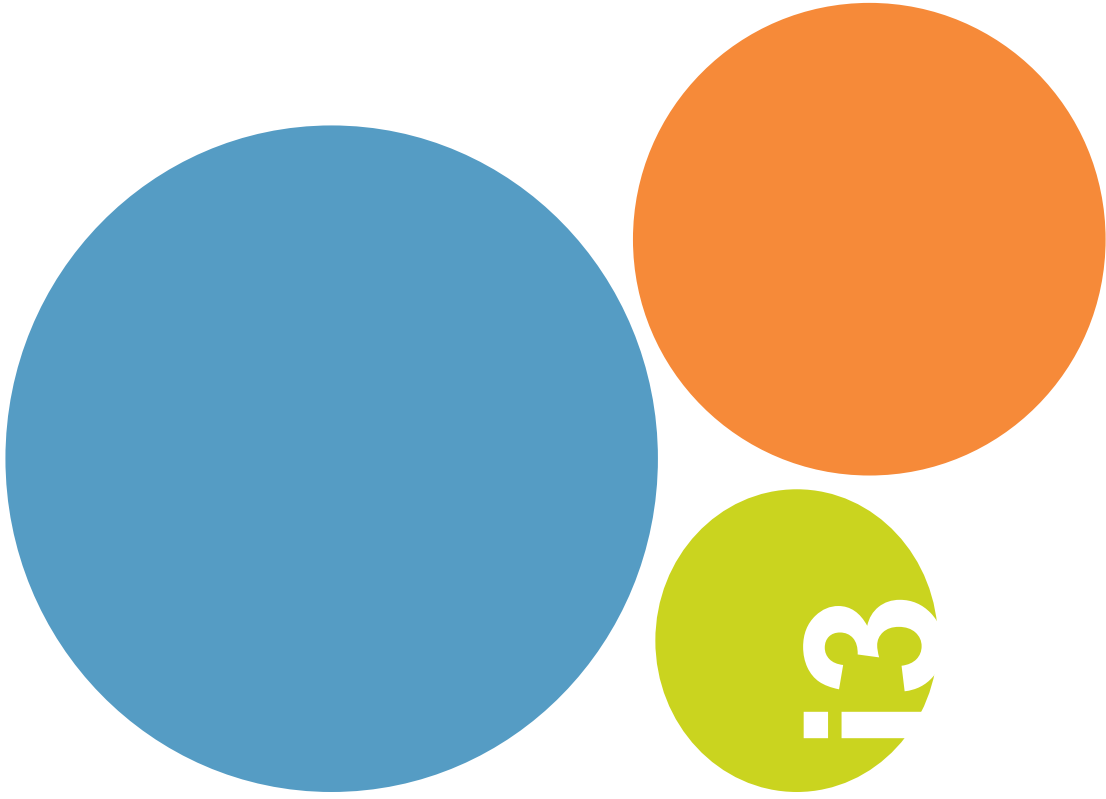
**Multiple Voice Codec:
Rec. Voice Codecs
Guidelines for FoIP Interc.**

**Signalling:
SIP, SIP-I
ISUP/SIP, SIP-I mapping**

**Testing Procedure:
Migration Interc. Form
Interop. Test Plan**

**Security:
White Paper in 2011**

Transitioning the Industry to IP



Thank You!

Go to www.i3forum.org