C&I Training on Conformity and Interoperability

ITU-D

Jun/2015
1. Next-generation Networks (NGN) basic concepts. Integration testing – interoperability aspects. ITU-T Q.3909 and ITU-D Q26/2 Report

2. SIP Protocol

3. H.248 / Sigtran Protocol

4. NGN Lab Instrumentation; Protocols: SIP

5. NGN Lab Instrumentation; Protocols: H.248/Sigtran
Next-generation Networks (NGN) basic concepts. Integration testing – interoperability aspects. ITU-T Q.3909 and ITU-D Q26/2 Report
TDM Network
Advantages of circuit-switched phone networks:
- Capillarity
- QoS – Quality of Service
- Real-time voice optimized network

Disadvantages of circuit-switched phone networks:
- Resource monopolization
- Hierarchical network
- Closed systems supplied by limited vendors
- Specialized network
- Low growth rate
Animation of an PSTN (TDM)
Phone call
Network Convergence
Data networks based on TCP/IP Protocol began in the late 60's when the first networks have emerged through the ARPANET, "joining" North American universities.

Advantages:
- Optimized resource usage
- Not hierarchical
- Multi-service Network
- Steady year-long growth

Disadvantages:
- In the past - no quality of service (QoS) guarantee
- Nowadays – some QoS guarantee

It is possible to talk over IP Networks.
The explosion of data transfer has revolutionized the telecommunication environment, especially the Internet (www, FTP, email, Facebook, Twitter, WhatsApp, Skype), indicating a shift from a “Voice Based” network to a "Data Based" network.

Thus, the Multi-Service Network emerges!
Network Convergence
High level view of the converged environment

Always on with any devices
Anytime, anywhere and in any form
Voice and multimedia
Self service, intuitive
Simple for the end user
Secure, trusted and reliable
Single data-based telecommunications networks providing high-quality integrated voice, video and data services, strategically offering multi-media based services and minimizing costs for both providers and consumers.

- **Factors propelling convergence:**
  - *Vigorous data traffic growth*
  - *Technological factors*
    - Statistical multiplexing
    - Protocols and networks such as IP/MPLS that transport media with guaranteed quality of service.
    - Routers, computers, gateways and other elements are simpler, more economical and easier to manage than TDM telephone network equipment.
    - Codecs: high-quality voice/video compression and packing.
ITU-T Recommendation Y.2001

"A packet-based network able to provide telecommunication services and able to make use of multiple broadband, QoS enabled transport technologies and in which service-related functions are independent from underlying transport-related technologies. It enables unfettered access for users to networks and to competing service providers and/or services of their choice. It supports generalized mobility which will allow consistent and ubiquitous provision of services to users."
It is possible to talk over IP Networks

Signaling takes typically different path than media does
NGN basic concepts

- Packet-based transfer
- Separation of control functions among bearer capabilities, call/session, and application/service
- Decoupling of service provision from transport, and provision of open interfaces
- Support for a wide range of services, applications and mechanisms based on service building blocks (including real time/streaming/non-real time and multimedia services)
- Broadband capabilities with end-to-end QoS (Quality of Service)
- Interworking with legacy networks via open interfaces
NGN basic concepts

- Generalized mobility
- Unfettered access by users to different service providers
- Unified service characteristics for the same service as perceived by the user
- Converged services between fixed/mobile
- Independence of service-related functions from underlying transport technologies
- Support of multiple last mile technologies
- Compliant with all regulatory requirements, for example concerning emergency communications, security, privacy, lawful interception, etc.
NGN Architecture

NGN basic concepts

**TDM Architecture**
- IN
  - Services
    - Call Control Switching
      - Switching Subscriber A/D Trunk

**NGN Architecture**
- Service Platform NGIN
  - SIP
    - Session Control
      - H.248 MGCP SIP
    - Transport Media User
• Responsible for services offered to users connected to the network

• Responsible for defining, supervising, clearing and rating calls over the IP network. The functions of this layer are:
  — Signaling
  — Call Control
  — Service Layer Access

• This is the layer where the media servers are, responsible for:
  — Processing media
  — Message recording and playback
  — Text-to-speech conversion and speech recognition
NGN – Softswitch Approach

Service
- IN
- Billing
- NMS
- Media Server
- Application Server

Control

Core

Access
- SG
- MGW
- IAD
- SBC
- LMG
- PSTN
- PRI / R2
- POTS / IP
- Private Network
- MSAN
- IP
NGN
NGN basic concept

- SS7 message conversion to the IP network (SIGTRAN).
- Establishes and terminates IP-SS7 connections. Keeps the connection status between networks.
- Establishment and termination of sessions in IP network.
- Maintains the state of all calls and the resources allocated to them.
- Send/receive signals and messages to/from other networks and protocols.
- Media adaptation between networks.
- Protocol conversion between IP and TDM.
- Create and delete protocol entities.
- MGC provides the QoS information.
User needs

- Higher-speed Internet
- Low cost
- Transparency between wired and wireless technology
- Low-cost apps with user-friendly interfaces

Service Provider Goals

- Easily manageable networks
- Planned networks
- Services independent of networks
- Networks that can evolve
- Simple, open user access
- Create open, fast platforms
- Guaranteed network QoS

NGN – to remember

“A proposal to evolve current voice-based telecommunication networks to data-based networks, integrating voice, video and data, combining wired and wireless technologies, and giving users unlimited, uninterrupted access with support for mobility.”
NGN
Class V

Class V SofftSwitch basic call
NGN architecture and technologies
NGN architecture and technologies
Signalling Gateway
Specification process for NGN conformance testing and interoperability testing

1. Define the target (service, application, protocol, function)
2. Define specifications (Protocol, Service, QoS)
3. Define test specifications to examine tests
4. Build up test bed and examine tests

Figure 1 – Typical NGN conformance and interoperability test specification process
Points and Protocols to test

Most requested tests

<table>
<thead>
<tr>
<th>Protocol</th>
<th>Test type</th>
<th>Who request</th>
</tr>
</thead>
<tbody>
<tr>
<td>SCC#7 - ISUP</td>
<td>Conformance</td>
<td>Service Provider</td>
</tr>
<tr>
<td></td>
<td>Performance (Traffic)</td>
<td>Anatel</td>
</tr>
<tr>
<td>Sgtran M3UA</td>
<td>Conformance</td>
<td>Service Provider</td>
</tr>
<tr>
<td>H.248</td>
<td>Conformance</td>
<td>Service Provider</td>
</tr>
<tr>
<td>SIP-I</td>
<td>Conformance</td>
<td>Service Provider</td>
</tr>
<tr>
<td>SIP / H.323</td>
<td>Conformance</td>
<td>Service Provider</td>
</tr>
<tr>
<td>ISDN-PRI</td>
<td>Conformance</td>
<td>ANATEL</td>
</tr>
<tr>
<td></td>
<td>Manufacturer</td>
<td>Manufacturer</td>
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<tr>
<td></td>
<td>(consulting services)</td>
<td>(consulting services)</td>
</tr>
<tr>
<td>SIP</td>
<td>Performance (Traffic)</td>
<td>Service Provider</td>
</tr>
<tr>
<td></td>
<td>Voice Quality</td>
<td>ANATEL</td>
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<tr>
<td></td>
<td>Interoperability, SIP-ISUP</td>
<td>ANATEL</td>
</tr>
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<td></td>
<td>Manufacturer</td>
<td>Manufacturer</td>
</tr>
<tr>
<td></td>
<td>(consulting services)</td>
<td>(consulting services)</td>
</tr>
</tbody>
</table>
Test Activities

Where to measure?
What to measure?
How to measure?

IP NETWORK

Protocols

- Sigtran
- H.323 or SIP
- MEGACO
- MGCP

Testing Methods

- Functional
- Interoperability
- Conformance
- Interworking
- Performance
Test Activities

**Protocols**

**International standards**
ITU-T, ETSI, IETF, ...

**Documents**
Tests Procedures

**Other documents**
Profiles

- Sigtran
- H.323 or SIP
- H.248

**What to measure?**

- Tester

**Where to measure?**
What to Measure?

- Measurement of messages and parameters of NGN protocols in the IP network.
- Performance: signaling and voice traffic
- Evaluation of quality and intelligibility of voice

Where to Measure?

- Choice of adequate test system
- Localization of the points of measurement
- Definition of the testing methods
Conformance / Interoperability Tests

• Advantage
  — Ensure equipment network performance
  — Ensure interoperability with equipment and systems from discrete vendors
  — Quality Assurance:
    ▪ Ensure equipment meets the technical requirements needed to provide services
    ▪ Ensure elementary problems will not occur
Testing Methods

- **Functional Testing**
- **Performance Testing**
- **Interoperability Testing**
- **Interworking Testing**
- **Conformance Testing**
Testing Methods

Test System

System Under Test

- Advantage
  - The Test System simulates the real network element (without needing the actual element) with the involved protocols, e.g., MGW, MGC, PSTN

- Case
  - Service Provider
  - Basic calls and functionalities

- Functionalities/features are tested

- Test system and/or real equipment (terminal) is used (Real Equipment Test)
• System under test is stimulated by a huge number of simulated end users.

• Scope of test is stability under high load.

• No functional verification.

**Advantage**
- Generates heavy traffic (by simulating many terminals).

**Case**
- High Service provider traffic
- MOS tests for service providers

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**Testing Methods**

- **Functional Testing**
- **Performance Testing**
- **Interoperability Testing**
- **Interworking Testing**
- **Conformance Testing**
• Verify that equipment from 2 different vendors can work together (can interwork)

• Verification of all protocols of a dedicated interface

**Advantage**
— To show that two or more implementations will cooperate to provide the specified end-to-end functions

**Case**
— Service Provider ISUP tests
Test that features and services defined in protocol A are correctly mapped to features and services of protocol B.

- **Advantage**
  - Shows the ability of dissimilar protocols (such as ISDN and GSM) to exchange service information.

- **Cases: Anatel**
  - ISUP-ISUP
  - SIP-RDSI
  - ISUP-transit
Testing Methods

Test System

- Test of one dedicated protocol layer within the system under test
- Conformance to the underlying standard (ETSI, ITU-T,...) is verified

System Under Test

- **Advantage**
  - Shows that a particular implementation complies with the protocol requirements specified in the associated base standard

- **Case**
  - Anatel
  - Service Providers
  - Vendors (consulting)
ITU-T Conformance Testing
Anatel Certification
Conformance testing for the NGN procedure:

• Preparation for testing
  — Set the test object, target interface and target Recommendations
  — Set the physical configuration and target product
  — Define the test scenarios

• Test operations
  — Static conformance review
  — Test selection and parameterization
  — Test campaigns (examine conformance testing according to the scenarios)
  — Analyze test output

• Test report production
Usual test workflow

Tester Assigned → Requirements Shared

Prepare Lab ← Plan test

Execute tests ← Check defects

Success → Unsuccessful

Problem unsolved

Complete
ANATEL Certification
Steps for lab deployment

• Conformance test equipment adherent to pertinent standards

• Equipment (System under test) comprising all functionalities
  — Team specialized in equipment

• Test specialist
  — Know tests norms
  — Know specifications of protocols involved in tests
  — Know how to configure and operate test equipment
  — Know setup
  — Monitor results
  — Save test result logs

• The Specialist evaluates test results
  — If the results ARE NOT as expected, a failure report is sent, indicating the item not compliant with that specific standard

• After tests are run, the specialist generates a report for the customer
  — DCB, Operator, Manufacturer, Vendor
ANATEL Certification Process

Steps for telecommunications products / ICT checking and approval in Brazil

1. Manufacturer (national) or representative chooses a Designated Certification Organism (OCD) and supplies all product technical information

2. OCD analyses the product and features and determines the standards and applicable tests

3. Manufacturer (national) or representative chooses a laboratory and supplies a product sample

4. Laboratory performs the tests laid down in the regulation and issues the test report

5. OCD analyzes the test results, issues the certificate of conformity and register it on Anatel’s Management and Homologation Certification System (SGCH)

6. Anatel analyses all documentation and issues the Homologation Certificate
Protocols

H.323
SIP
H.248/MGCP
SIGTRAN
What is a Protocol

• Protocol is a convention that enables and controls connections, communication, and data transfer between two computer systems.

• A protocol can be defined as "the rules that govern" the syntax, semantics and synchronization of communication.

• Protocols can be implemented by hardware, software or a combination of the two.
H.323
H.323 Architecture

Tests for ANATEL

ETSI TS 101 804-2: "Methods for Testing and Specification (MTS); Conformance Test"

Specification for ITU-T H.225.0 (Terminal, Gatekeeper and Gateway); Part 1: Protocol Implementation Conformance Statement (PICS) proforma".


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Audio Apps
- G.711
- G.729
- G.723.1
- H.261
- H.263

Video Apps
- RTP
- RTCP
- H.225.0 RAS
- H.225.0 Call signaling
- H.245 Control signaling
- T.120 Data

Terminal control and management

Physical, DataLink
- UDP
- TCP
- IP
SIP - Session Initiation Protocol
Session Initiation Protocol
SIP Architecture

- **MGC**
- **AS**
- **Registrar**
- **Redirect**
- **Proxy**
- **SIP Servers**
- **Terminal**
- **MGW**
- **SGW**
- **H.248**
- **SIP-I**
- **SIGTRAN**

Connections:
- SIP
- SIP-I
- SIGTRAN
- RTP

Network Components:
- **Session Initiation Protocol** (SIP)
- **Signaling System 7** (SS7)
- **Real-Time Protocol** (RTP)
- **H.248**
- **AS** (Application Server)
- **MGW** (Media Gateway)
- **SGW** (Signaling Gateway)
Session Initiation Protocol
SIP basics

- Approved by IETF (Internet Engineering Task Force) by RFC 2543, in March 1999.

- Used to create, modify and terminate multimedia sessions, providing the means for determining members address and location.

- It is a client/server protocol, similar to HTTP (Hypertext Transfer Protocol – RFC 2068) in terms of syntax and semantics.
  
  — Requests are generated by client entity and sent to the server. The server processes the message and sends a response to the client.
• In a SIP network, with specific functions, there are three kinds of servers

— *SIP Registrar*
  - Receive registration requests from users
  - Maintains user’s location at a specific server (like HLR or HSS)

— *SIP Proxy Server*
  - Relays call signaling
  - Transparent to end-devices
  - Allows for additional services (call forwarding, forking, etc.)

— *SIP Redirect Server*
  - Redirects callers to other servers

All of these elements are logical and could be part of a single server!
In SIP operation

- All messages have an initial line specifying the method and the protocol, and some may contain an optional body, describing the section.

Two types of messages are used in SIP

- Request
  - Contains methods indicating the requested action.

- Response
  - Based on the HTTP structure. There are six kinds of response messages.
Session Initiation Protocol
SIP basics

• SIP – Request Messages
  — Invite: Invites a user to participate in a call and establishes a connection.
  — Bye: Terminates a connection.
  — Options: Requests information about the capabilities of a caller.
  — ACK: Indicates an accepted "invite".
  — Register: Informs user location to server.

• SIP – Responses
  — 1XX: Progress
  — 2XX: Successful Request
  — 3XX: Redirection
  — 4XX: Incorrect Request
  — 5XX: Server Failure
  — 6XX: Global Failure
Session Initiation Protocol
Call Flow Example

SIP Proxy
Redirect

Terminal

REGISTER
200 OK

INVITE
100 Trying

180 Ringing
200 OK

ACK

180 Ringing
200 OK

ACK

BYE
200 OK
• Standard: IETF RFC 3261: "SIP: Session Initiation Protocol"

• Standard Test
  — Sub groups may be subdivided in three subgroups:
    ▪ Valid behaviour (V)
    ▪ Invalid behaviour (I)
    ▪ InOpportune behaviour (O).
  — Conformance test required by Anatel (Terminals)
• **EX1: 5.3.1.1.1 Valid Behaviour**
  
  — **TPId:** SIP_CC_OE_CE_V_001
  
  — **Status:** Mandatory
  
  
  — **Purpose:** Ensure that the IUT, to establish a call sends an INVITE request including at least To, From, CSeq, Call-ID, Max-Forwards, Contact and Via headers.

• **EX2 5.5.3.1 Valid Behaviour**
  
  — **TPId:** SIP_MG_OE_V_007
  
  — **Status:** Mandatory
  
  — **Ref:** RFC 3261 [1] section 7.3.1.
  
  — **Purpose:** Ensure that the IUT, when an INVITE client transaction is in the Calling state, on receipt of a Success (200 OK) response including a headers set with short names, sends an ACK request.
The following procedures will take place in the lab:

- Conformance Tests for SIP
- Tests involving the so-called basic network:
  - *Success case study*
  - *Failure case study*
MGCP - Media Gateway Control Protocol
H.248 / MEGACO
Media Gateway Control Protocol
Basics

• Media Gateway Protocol
  — Defined by IETF (Internet Engineering Task Force, in RFC 2705).

• Ideal proposal for building IP networks with voice and data convergence
  — Designed with a distributed architecture, where operational, signaling and media control issues are handled separately.

• Main objective
  — Allow all the gateways of an IP network (Media Gateways) to be controlled by an external agent, called Media Gateway Controller or Call Agent/Softswitch.
Main features

- Jointly developed by the IETF and ITU-T
- Protocol Architecture for control between gateways and other elements of a multimedia system
- Add point-to-point interoperability aspects
- It must be used in conjunction with H.323 or SIP
- Assumes a centralized intelligence model on the network
- Emphasizes interoperability

Protocol architecture for Gateway control

- MGCP style
  - Control between Gateways and other elements of a multimedia system and its controllers - MGC
H.248/MEGACO – Main messages

• **Add**
  — *Adds a termination to a context.*

• **Modify**
  — *Modifies a termination's properties, events and signals.*

• **Subtract**
  — *Disconnects a termination from a context, returning statistics of the termination's participation in the context. The Subtract command in the last termination in context deletes the context.*

• **Notify**
  — *Allows the Media Gateway to inform the Media Gateway Controller of events taking place in a specific termination.*
H.248/MEGACO – Protocol Commands

• **MGC → MG:**
  - **Add, Subtract** - Add or Remove terminations in a context
  - **Modify** – Modify features (Properties, Events and Signals) of a termination
  - **Move** - Move a termination to other context
  - **Audit Values** – verify MG/Termination parameters values
  - **Audit Capabilities** – verify MG/termination features
  - **ServiceChange** - communicates a state change element

• **MG → MGC:**
  - **Notify** - Warns events detected by MG
H.248/MEGACO – Protocol Responses

- Basically, two situations occur:
  
  - *If the transaction requested through a command completes successfully, the receiver returns the same “transaction ID” in a Reply.*
  
  - *If the transaction requested through a command does not complete successfully, the receiver returns the same “transaction ID” in a reply stating the real cause for failure, with responses such as:*

    - 400 – Bad request
    - 401 – Protocol error
    - 500 – Internal gateway error
    - 502 – Not ready
    - 503 – Service not available
    - 581 – Nonexistent
H.248/MEGACO
Context, termination and association

Media Gateway
H.248/MEGACO
Call setup (1/2)

MGW B

Null Context

T4

ADD T4 to new context

MGW A

Null Context

T1

ADD T1 to new context
ADD T4 to new context and create termination (T3) and ADD to it.

ADD T1 to new context, create termination (T2) and ADD to the same context.
H.248/MEGACO

Call setup (1/2)

Context =${Add=T1, Add=${mode=receiveonly}}

Reply{Context=1001{Add=T1,Add=T2{LocalDescriptor}}}

Context=${Add=T4, Add=${RemoteDescriptor}}

Reply{Context=2002{Add T4,Add=T3{LocalDescriptor}}}

Modify=1001{Modify=T2{Mode=sendreceive,RemoteDescriptor}}

Reply{Context=1001{Modify=T2}}
H.248/MEGACO
Call setup (2/2)
MEGACO / H.248 Class 4 call flow

Obs.: There are Reply messages, not represented in draw
• **Standard: ITU-T Recommendation H.248**

• **Standard Test**
  
  — **ETSI TS 102 374-2 Methods for Testing and Specification (MTS) - Conformance Test**
  
  Specification for ITU-T H.248.1

  — **Protocol groups - The protocol groups identify the two roles of the IUT (Implementation Under Test):**

  ▪ Media Gateway (MG) and
  
  ▪ Media Gateway Controller (MGC) as defined in ITU-T Recommendation H.248.1 [1].

• **Conformance test required by Vendors**
EX1: 5.2 TPs for Media Gateway (MG)

5.2.1 Procedures using Add command (AD)

5.2.1.1 Valid behaviour test purposes (BV)

TP/MG/AD/BV-01 Reference: ITU-T Recommendation H.248.1 [1], clause 7.2.1

Initial condition: any

Ensure that the IUT, on receipt of a Transaction Request containing:

Action request with:
- CID set to CHOOSE;
- ADD Command request with: TID set to CHOOSE;

NOTE: e.g. for creation of a RTP Termination. Sends a Transaction Reply containing:

Action reply with:
- CID set to a specific value (assigned by the MG);
- ADD Command reply with: TID set to a specific value (assigned by the MG).
EX2: 5.3 TPs for Media Gateway Controller (MGC)

5.3.1 Procedures using Add command (AD)

5.3.1.1 Valid behavior test purposes (BV)

TP/MGC/AD/BV-01 Reference: ITU-T Recommendation H.248.1 [1], clause 7.2.1

Selection criteria:

- Initial condition: any

- Ensure that the IUT, in order to create an ephemeral termination in a new context, sends a Transaction Request containing:

  - Action request with:
    - CID set to CHOOSE;
    - ADD Command request with: TID set to CHOOSE
H.248 LAB Scenario
The following procedures will take place in the lab:

- Conformance Tests for H.248
- Tests involving the so-called basic network:
  - Success case study
  - Failure case study
SIGTRAN
• Bridge between PSTN and NGN networks
• Carries SS7 application signalling in IP networks
• SS7 over IP is known as SIGTRAN
• There are 4 types of SGs
  — **M3UA**: Transfer information through the ISUP SS7 nodes; It can also be used for transaction-based services (such as Global Title Translation) by the SCC over the M3UA gateway.
  
  — **SUA**: Used to transfer transaction information (such as database lookup) through SS7 nodes.
  
  — **M2UA/M2PA**: Used to transfer SS7 links through SS7 nodes; at this point the SS7 element in IP network tipically becomes to equivalent IP-STP.
  
  — **IUA**: Used to transfer RDSI (Q.931) information over IP.
SIGTRAN

MGC

IP

SIGTRAN

MEGACO

SG

MGW

SS7

Voice

PSTN

PSTN

SS7
CPqD experiences

CASE

• Consulting Services to Service Provider:
  — *SIP conformance tests*
  — *SIP traffic tests*
  — *ISUP and SIGTRAN (M3UA, M2PA, SCTP) conformance tests*

• ANATEL Tests:
  — *SIP conformance tests*

• Manufacturer development tests:
  — *SIP conformance tests*

• On service provider tests:
  — *SIP compliance tests*
1. Migration from existing networks to NGN networks

2. Interoperability aspects. SIP-I migration from existing networks to NGN (SIP-ISUP (Q.784/Q.850))

3. Quality (PESQ recommendation P.862)

4. Lab interoperability aspects. SIP-ISUP SIP-I (Q.1912.5 Profile C)

5. Lab Voice Quality (PESQ recommendation P.862)
Migration from existing networks to NGN
Migration from existing networks to next-generation networks for developing countries:

— technical,

— regulatory and

— policy aspects

Motivations to migrate from legacy network infrastructure to the new network infrastructure.
Migration to NGN
Telecom voice revenue

Fixed telephony gross revenue

Consumer Spending: Telecom

% Change in Voice Revenue (Fixed Lines) - 5 Year CAGR

Source: TeleGeography

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Migration to NGN
Following the trend caused by the business flows

- Reducing revenue of fixed voice services
- Requesting more mobile oriented services and IP based capabilities over fixed and/or mobile broadband

Voice services from legacy fixed based to mobile and IP based
Migration to NGN
Total comparator country retail telecoms revenue: 2008 to 2013

Retail fixed-line voice revenues: 2008 to 2013

Sources: IHS / industry data / Ofcom
Migration to NGN
Following the trend caused by the business flows

For the compensation of revenue reduction

- cost reductions by sharing network infrastructure
- cost reductions by sharing systems

- Requirements to the reduction of network and service infrastructure deployment cost:
  - Reduced OPEX and enhancing streamline operations.
  - Integrated platforms for provisioning of various types services and applications.
  - Integrated operation platforms including integrated maintenance and training.
  - Centralized Management and Control.
Migration to NGN
Operator’s Challenges on Migration

• Support of business continuity required to maintain ongoing dominant services and customers that require carrier-grade service.

• Flexibility to incorporate existing new services and react quickly to the ones that appear on real time.

• Profitability to allow feasible return on investments and in the best practices market values.

• Survivability to allow service assurance in case of failures and external unexpected events.

• Quality of Service to guarantee the Service Level Agreements for different traffic mixes, conditions and overload.

• Interoperability across networks to allow the carrying of end to end services for flows in different network domains.
Migration to NGN
Considerations: Signaling and Control

• PSTN/ISDN uses signalling systems such as
  — Analogue line signalling
  — Channel associated signalling (CAS) like signalling systems R1 [Q.310-Q.332], R2 [Q.400-Q.490]
  — Common channel signalling (CCS), like SS7 or DSS1 [Q.931].

• All these signalling systems are for the circuit switched networks.
• Since NGN transport is packet-based, other suitable types of signalling (e.g., SIP-I [Q.1912.5], etc.) may be required.
• Since the NGN has to work with the PSTN/ISDN and other networks, interworking between NGN signaling systems and the legacy network signalling systems is required.
Issues for NGN design

- New models needed to represent multiservice flows
- New dimensioning methods for resources handling multimedia services with QoS
- New measurement procedures for aggregated multi-service traffics
- New procedures to ensure interoperability and end-to-end performance across multiple domains
- Redefinition of network segments at the new structure and for QoS quota assignment
- New units to define dimensioning and costing for interconnection
Migration to NGN
Changes from current scenario towards target network

Source: ITU
Migration to NGN

Topology reconfiguration for Core

Softswitches /MGCs in few sites

Trunking gateway in each regional site

Packet mode network

IP links short distance

Regional Level

LEX Layer

Local Exchanges

Remote Units

Source: ITU
Migration to NGN
Migration strategy - Core

• Dominated by high capacity and protection level

  — Overlay deployment for full coverage in all regions
  — Quick deployment needed for homogeneous end to end connections (2 to 3 years)
  — Strong requirements for high quality, protection and survivability
  — Importance of the optimization for location and interconnection

Source: ITU
Migration to NGN
Topology reconfiguration for Core

LEX Layer

IP links Long distance

MGCs located in few sites

Trunking gateway in each local site

Regional layer

Packet mode network

Local Exchanges

Remote Units

Source: ITU
Migration to NGN
Local / Edge level migration: grow with NGN

Source: ITU
Dominated by functions migration investment and interoperability

— Move from joint switching and control to separated control and media GW

— Introduce Multimedia Services at all areas

— Optimize number, location of nodes and interfaces among existing and new network

— Requires longer time and higher investments due to variety of geo-scenarios and geographical distribution

Source: ITU
Migration to NGN
Local / Edge level migration: grow with NGN

Rest of the TDM network

Trunk gateways

Packet mode network

IP links

Access gateways

Softswitches/MGCs

Access gateway

Obsolete TDM LEXs

Access equipment

Source: ITU
SIP-I migration from existing networks to NGN SIP-ISUP (Q.784/Q.850)
SIP-I migration from existing networks to NGN SIP-ISUP (Q.784/Q.850)

- SIP-ISUP (Q.784/Q.850)
- Q.784 Describes isup conformance tests standard
- Q.850 - Release cause map
- RFC 3398 e Q.1912.5 - mappings of Q.850 to SIP
The Release Causes mapped in Q.850 are very important for gap identification in operators (KPI), which check Quality of Service indicators requested by Anatel

Some of these causes are part of the ISUP conformance tests (Q.784)

The SIP fault mapping (4xx) contains REASON CAUSE map with the Q.850

Ex. 486 busy here (REASON CAUSE #17 user busy)
### Table 21/Q.1912.5 – Receipt of the Release message (REL)

<table>
<thead>
<tr>
<th>← SIP Message</th>
<th>← REL Cause Indicators parameter</th>
</tr>
</thead>
<tbody>
<tr>
<td>404 Not Found</td>
<td>Cause Value No. 1 (&quot;unallocated (unassigned) number&quot;)</td>
</tr>
<tr>
<td>500 Server Internal Error</td>
<td>Cause Value No. 2 (&quot;no route to network&quot;)</td>
</tr>
<tr>
<td>500 Server Internal Error</td>
<td>Cause Value No. 3 (&quot;no route to destination&quot;)</td>
</tr>
<tr>
<td>500 Server Internal Error</td>
<td>Cause Value No. 4 (&quot;Send special information tone&quot;)</td>
</tr>
<tr>
<td>404 Not Found</td>
<td>Cause Value No. 5 (&quot;Missed call trillion prefix&quot;)</td>
</tr>
<tr>
<td>500 Server Internal Error (SIP-I only)</td>
<td>Cause Value No. 8 (&quot;Preemption&quot;)</td>
</tr>
<tr>
<td>500 Server Internal Error (SIP-I only)</td>
<td>Cause Value No. 9 (&quot;Preemption-circuit reserved for reuse&quot;)</td>
</tr>
<tr>
<td>480 Busy Here</td>
<td>Cause Value No. 17 (&quot;user busy&quot;)</td>
</tr>
<tr>
<td>480 Temporarily unavailable</td>
<td>Cause Value No. 18 (&quot;no user responding&quot;)</td>
</tr>
<tr>
<td>480 Temporarily unavailable</td>
<td>Cause Value No. 19 (&quot;no answer from the user&quot;)</td>
</tr>
<tr>
<td>480 Temporarily unavailable</td>
<td>Cause Value No. 20 (&quot;subscriber absent&quot;)</td>
</tr>
<tr>
<td>480 Temporarily unavailable</td>
<td>Cause Value No. 21 (&quot;call rejected&quot;)</td>
</tr>
<tr>
<td>410 Gone</td>
<td>Cause Value No. 22 (&quot;number changed&quot;)</td>
</tr>
<tr>
<td>No mapping</td>
<td>Cause Value No. 23 (&quot;redirection to new destination&quot;)</td>
</tr>
<tr>
<td>480 Temporarily unavailable</td>
<td>Cause Value No. 25 (&quot;Exchange routing error&quot;)</td>
</tr>
<tr>
<td>502 Bad Gateway</td>
<td>Cause Value No. 27 (&quot;destination out of order&quot;)</td>
</tr>
<tr>
<td>484 Address Incomplete</td>
<td>Cause Value No. 28 (&quot;Invalid address format (address incomplete)&quot;)</td>
</tr>
<tr>
<td>500 Server Internal Error</td>
<td>Cause Value No. 29 (&quot;facilty rejected&quot;)</td>
</tr>
<tr>
<td>404 Not Found</td>
<td>Cause Value No. 91 (&quot;invalid state selection&quot;)</td>
</tr>
<tr>
<td>500 Server Internal Error</td>
<td>Cause Value No. 95 (&quot;invalid message, unspecified&quot;)</td>
</tr>
<tr>
<td>500 Server Internal Error</td>
<td>Cause Value No. 97 (&quot;Message type present or not implemented&quot;)</td>
</tr>
<tr>
<td>500 Server Internal Error</td>
<td>Cause Value No. 99 (&quot;information element/parameter present or not implemented&quot;)</td>
</tr>
</tbody>
</table>
Interworking of ISUP with SIP using Profile C (SIP-I)-ITU-T Rec. Q.1912.5

NOTE 1 – Any SIP entity along the signalling path to the I-IWU, or the I-IWU itself, may return a 100 Trying provisional response either by configuration or because it determines that a further response will take longer than 200 ms to generate. This is a purely SIP matter with no interworking significance, but is depicted for realism in this and subsequent figures.

NOTE 2 – ACM contained the following indicators:
Called Party Status = “subscriber free”, ISDN Access Indicator = “ISDN access”
Unsuccessful call set-up procedures and call flow diagrams for basic call control

Figure II.8/Q.1912.5 – Unsuccessful basic call set-up from ISUP to SIP
II.5.2.3 Normal call release procedure

Figure II.9 shows a normal call release procedure initiated from the ISUP side of the call. This call flow assumes that no resource reservation teardown signalling is required on the SIP side of the call.

Figure II.9/Q.1912.5 – Normal call release from ISUP to SIP
SIP-I LAB Scenario
Quality of Service (QoS)
QoS in Data Network

- The packet and circuit networks were designed with different goals:
  - Data network (critical, non-critical)
  - Telephony network (ToIP, VoIP)

- To guarantee Quality of Service for voice traffic on IP networks is the biggest challenge for the full integration of data networks and telephony.

Audio Quality

- Largely depends on codec and
- echo cancellation in use
<table>
<thead>
<tr>
<th>Service</th>
<th>Bandwidth (downstream)</th>
<th>QoS Requirement</th>
</tr>
</thead>
<tbody>
<tr>
<td>Broadcast TV (MPEG-2)</td>
<td>2 to 6 Mb/s</td>
<td>Parameterized</td>
</tr>
<tr>
<td>HDTV (MPEG-4)</td>
<td>6 to 12 Mb/s</td>
<td>Parameterized</td>
</tr>
<tr>
<td>PPV or NVoD</td>
<td>2 to 6 Mb/s</td>
<td>Prioritized</td>
</tr>
<tr>
<td>VoD</td>
<td>2 to 6 Mb/s</td>
<td>Prioritized</td>
</tr>
<tr>
<td>Picture in Picture (MPEG-2)</td>
<td>up to 12 Mb/s</td>
<td>Parameterized</td>
</tr>
<tr>
<td>PVR</td>
<td>2 to 6 Mb/s</td>
<td>Prioritized</td>
</tr>
<tr>
<td>Interactive TV</td>
<td>up to 3 Mb/s</td>
<td>Best effort</td>
</tr>
<tr>
<td>High speed Internet</td>
<td>3 to 10 Mb/s</td>
<td>Best effort</td>
</tr>
<tr>
<td>Video Conferencing</td>
<td>300 to 750 Kb/s</td>
<td>Prioritized</td>
</tr>
<tr>
<td>Voice/Video Telephony</td>
<td>64 to 750 Kb/s</td>
<td>Prioritized</td>
</tr>
</tbody>
</table>
Requirements for IP Telephony

- *Voice transmission in real time - latency <300 ms*

- **Signaling Procedure**
  - *Call Establishment*
  - *Call Control*
  - *Additional services*

- *Public switched systems and mobile telephony interfaces*

- **QoS guarantee**
  - *Compression Techniques*
  - *Silence suppression*
  - *Echo control*
Voice packing - codecs

- **Traditional systems - Voice**
  - 4 kHz band
  - 8kHz sampling
  - Each voice channel - 64kbit/s (8000 samples x 8 bits)
  - ITU-T Recommendation G.711 (PCM)

- **VoIP systems - bandwidth demand is critical**
  - Speech signal compression algorithms needed
  - Bandwidth savings
  - DSP - Digital Signal Processors

- Compression can take place with or without loss of quality. Everything depends on the degradation that is admitted to the signal and the compression factor that you want to achieve: G.711, G.726, G728, G.729
Codecs are used

- During the voice signal packing

- To reduce bandwidth voice compression techniques are used
  - The signal compression performed in encoders (codecs) are based on processing techniques to remove redundant information, predictable or useless.

  - Compression can take place with or without loss of quality.
  - Tudo depende da degradação que se admite para o sinal e do fator de compressão que se deseja atingir.
  - Everything depends on the admitted signal degradation and the compression factor that is wanted to achieve.
Quality of Service (QoS)

Voice over IP - Use of Data Networks that uses IP protocols (TCP / UDP / IP) for voice signal transmission in real time in the form of data packets.
RTP: Real Time Protocol

Animation of RTP

Protocol use
## Quality of Service (QoS)
### ITU-T Codecs

<table>
<thead>
<tr>
<th>Method</th>
<th>Rec ITU-T</th>
<th>Band (kbps)</th>
<th>MOS</th>
<th>Complexity</th>
<th>Delay (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td>PCM</td>
<td>G.711</td>
<td>64</td>
<td>4,1</td>
<td>-</td>
<td>0.75</td>
</tr>
<tr>
<td>ADPCM</td>
<td>G.726</td>
<td>32</td>
<td>3,85</td>
<td>Low</td>
<td>1</td>
</tr>
<tr>
<td>LD-CELP</td>
<td>G.728</td>
<td>16</td>
<td>3,61</td>
<td>Low</td>
<td>3-5</td>
</tr>
<tr>
<td>CS-ACELP</td>
<td>G.729</td>
<td>8</td>
<td>3,92</td>
<td>Medium</td>
<td>10</td>
</tr>
<tr>
<td>CS-ACELP</td>
<td>G.729a</td>
<td>8</td>
<td>3,7</td>
<td>Medium</td>
<td>10</td>
</tr>
<tr>
<td>MP-MLQ</td>
<td>G.723.1</td>
<td>6,3</td>
<td>3,9</td>
<td>High</td>
<td>30</td>
</tr>
<tr>
<td>ACELP</td>
<td>G.723.1</td>
<td>5,3</td>
<td>3,65</td>
<td>High</td>
<td>30</td>
</tr>
</tbody>
</table>
QoS considerations for VOIP

- Use G.711 CODEC when possible
  - Good Voice Quality
  - Bandwidth usually available in LAN and MAN

- Use G.729A or G.729AB to conserve bandwidth
  - Watch out for multiple transcodings
  - Be careful with VAD – subject to clipping effects

- QoS
  - IP networks do not guarantee that bandwidth will be available for voice calls unless QoS mechanisms are used
  - QoS to restrict delay, minimize packet loss
  - QoS techniques should be applied to support VoIP with acceptable, consistent and predictable voice quality
  - QoS mechanisms refer to packet tagging mechanisms and network architecture decisions on the TCP/IP network to expedite packet forwarding and delivery
Quality of Service (QoS) Parameters

- **QoS in Data Network**
  - QoS represents the set of techniques necessary to manage network bandwidth, delay, jitter, and packet loss.
  - QoS is defined by the minimum performance parameters which a network have to provide. These can be highlighted:
    - **Bandwidth (Flow)** - mechanism used by the network to share this bandwidth between several bandwidths offered by the application.
    - **Latency** - defines the retention time (delay) of a packet by the network equipment.
    - **Jitter** - time interval change between the arrival of packets which occurs by the random behavior of the network delay.
    - **Packet loss** - packets transmitted by the endpoint source that does not arrive at the destination endpoint.
There are several ways to deliver QoS, including the following:

**Network QoS Technologies**
- Ethernet 802.1Q/802.1p
- IP Differentiated Services (DiffServ)
- MPLS for Traffic-Engineered Paths

**VoIP Application QoS Technologies**
- Codec Selection
- VAD / Silence Suppression
- Call Admission Control / Bandwidth Management
- Packetization rate
- Jitter buffer size
• **MOS (Mean Opinion Score) - ITU-T P.800**

  — It is a technique based on human perception used to "measure" the quality of the voice understanding.

  — **There are two test methods:**

    ▪ Conversation opinion test
    ▪ Listening opinion test

  — The testers judge the quality of the voice transmission talking or listening to the voice samples, classified on a scale from 1 to 5 representing respectively:

    ▪ Bad, poor, fair, good and excellent.
    ▪ From 1 (bad) to 5 (excellent)
    ▪ Toll-quality > 4
Quality of Service (QoS)
PESQ

- **PESQ (Perceptual Evaluation of Speech Quality)**
  - PESQ - Perceptual Evaluation of Speech Quality
  - ITU-T improved the original model of MOS with Recommendation P.862 - PESQ for Narrow Band Codecs (3.1 KHz).
  - It is able to predict the subjective speech quality with a good correlation and a big variety of conditions, which may include coding distortions, errors, noise, filtering, delay and variable delay.

- It produces accurate predictions of quality in the presence of various behaviors during calls.
### Quality of Service (QoS)

#### Measuring QoE: MOS and the E-Model

- **Mean Opinion Score (ITU P.800)**
  - Subjective call quality measurement perceived by the user

- **E-Model (ITU G.107)**
  - Transmission planning tool for estimating user satisfaction
  - Objective measurement
  - E-model output: R value
    - Under 60 is not acceptable
    - Over 94.5 is unattainable in VOIP

<table>
<thead>
<tr>
<th>R-Value</th>
<th>User Satisfaction</th>
<th>MOS</th>
</tr>
</thead>
<tbody>
<tr>
<td>100</td>
<td>Very Satisfied</td>
<td>5.0</td>
</tr>
<tr>
<td>94</td>
<td>Satisfied</td>
<td>4.4</td>
</tr>
<tr>
<td>90</td>
<td></td>
<td>4.3</td>
</tr>
<tr>
<td>80</td>
<td>Some Users Dissatisfied</td>
<td>4.0</td>
</tr>
<tr>
<td>70</td>
<td>Many Users Dissatisfied</td>
<td>3.6</td>
</tr>
<tr>
<td>60</td>
<td>Nearly All Users Dissatisfied</td>
<td>3.1</td>
</tr>
<tr>
<td>50</td>
<td>Not Recommended</td>
<td>2.6</td>
</tr>
<tr>
<td>0</td>
<td></td>
<td>1.0</td>
</tr>
</tbody>
</table>
Quality of Service (QoS)  
Measuring QoE: MOS and the E-Model

<table>
<thead>
<tr>
<th>MOS</th>
<th>Quality</th>
<th>Impairment</th>
</tr>
</thead>
<tbody>
<tr>
<td>5</td>
<td>Excellent</td>
<td>Imperceptible</td>
</tr>
<tr>
<td>4</td>
<td>Good</td>
<td>Perceptible but not annoying</td>
</tr>
<tr>
<td>3</td>
<td>Fair</td>
<td>Slightly annoying</td>
</tr>
<tr>
<td>2</td>
<td>Poor</td>
<td>Annoying</td>
</tr>
<tr>
<td>1</td>
<td>Bad</td>
<td>Very annoying</td>
</tr>
</tbody>
</table>

There are many factors will affect PESQ score, such as:

1. Codec distortion
2. Packet delay and loss
3. Jitter butter delay
4. Background noise
Quality of Service (QoS)
G.114 – Determination of the effects of absolute delay by the E-model
Voice Quality LAB Scenario
Quality of Service (QoS)
Test scenario

- SIP-ISDN calls test scenario and voice quality measure
- Perceptual Evaluation of SpeechQuality (PESQ) in standard P.862
Thank you!

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augusto@cpqd.com.br

www.cpqd.com.br
Last, but not least
<table>
<thead>
<tr>
<th>S. No.</th>
<th>Protocol</th>
<th>ITU-T/IETF RFC No.</th>
<th>Title</th>
<th>Remarks</th>
</tr>
</thead>
<tbody>
<tr>
<td>3.</td>
<td>SIP-I</td>
<td>ITU-T Recommendation Q.1912.5</td>
<td>Inter working between Session Initiation Protocol (SIP) and Bearer Independent Call Control Protocol or ISDN User Part</td>
<td>Communication between two Soft switches</td>
</tr>
<tr>
<td>6.</td>
<td>RTP,RTCP</td>
<td>IETF RFC 3550, RFC3551</td>
<td>Real Time Protocol, Real Time Control Protocol</td>
<td>Delivery of Packetised Media Streams over IP</td>
</tr>
<tr>
<td>7.</td>
<td>Siganan</td>
<td>IETF RFC 2719</td>
<td>Framework Architecture for Signalling Transport</td>
<td></td>
</tr>
<tr>
<td>8.</td>
<td>SCTP</td>
<td>IETF RFC 2900</td>
<td>Stream Control Transport Protocol</td>
<td></td>
</tr>
<tr>
<td>9.</td>
<td>M2PA</td>
<td>IETF RFC 4165</td>
<td>MTP2 Peer to Peer Adaptation protocol</td>
<td></td>
</tr>
<tr>
<td>10.</td>
<td>M2UA</td>
<td>IETF RFC 3331</td>
<td>MTP2 User Adaptation Layer protocol</td>
<td></td>
</tr>
<tr>
<td>11.</td>
<td>M3UA</td>
<td>IETF RFC 4666</td>
<td>MTP3 User Adaptation Layer protocol</td>
<td></td>
</tr>
<tr>
<td>12.</td>
<td>SUA</td>
<td>IETF RFC 3868</td>
<td>SCCP user Adaptation protocol</td>
<td></td>
</tr>
<tr>
<td>13.</td>
<td>IUA</td>
<td>IETF RFC 4233</td>
<td>ISDN User Adaptation Layer Protocol</td>
<td></td>
</tr>
<tr>
<td>14.</td>
<td>V5UA</td>
<td>IETF RFC 3807</td>
<td>V5.2 User Adaptation Layer Protocol</td>
<td></td>
</tr>
</tbody>
</table>
Gostaríamos de deixar a mensagem:

- Importância da evolução das redes de circuitos para redes de pacotes
- Importância das normas ITU neste contexto
- Importância dos QOS redes VoIP
- Testes de interoperabilidade, garantem integração de equipamentos de fabricantes diferentes. As normas do ITU contribuem muito para isto.
- Release cause, protocolo ISUP, mapeia a causa das falhas nas redes (usada em operadoras, para KPI índices de completa mento de chamadas) presente também no protocolo SiP-(Reason cause da Q.850 do ITU)
- Facilidade de monitoração das novas redes ferramenta Free Wireshark
- Vantagens de construir um laboratório e que é necessário para compô-lo, um dos maiores ganho (know how), há dionivel no mercado vários equipamentos que contemplam as normas de testes de conformidade.
• **EXEMPLO DE 5 QUESTÕES**

  — O que é NGN,

  — Que tipo de testes pode ser realizado

  — Qual o ganho de realizar testes

  — Cite 2 protocolos

  — O que se mede em QoS
Migration to NGN
What to answer

• Questions and Answers

— *From where to start migration?*

— *Which topologies and connectivity are required?*

— *How network segments change in access, local and core?*

— *Which level of protection to assure?*

— *Where to locate new functionalities?*

— *How to ensure service continuity?*
The IMS specifications were developed for use with cellular access networks and were based on certain assumptions regarding the access network such as bandwidth available. Inherent differences between the different types of access networks will have concrete consequences on the IMS specifications.
Trends
3GPP/3GPP2 IMS Architecture

Source: IEEE
H.323 Call Flow Example

- Both endpoints have previously registered with the gatekeeper.
- Terminal A initiates the call to the gatekeeper. (RAS messages are exchanged).
- The gatekeeper provides information for Terminal A to contact Terminal B.
- Terminal A sends a SETUP message to Terminal B.
- Terminal B responds with a Call Proceeding message and also contacts the gatekeeper for permission.
- Terminal B sends an Alerting and Connect message.
- Terminal B and A exchange H.245 messages to determine master slave, terminal capabilities, and open logical channels.
MGCP / SIP-T call flow

Obs.: There are ACK in MGCP messages, not represented in draw
MGCP Class 4 call flow

A

PSTN

DTMF

IAM

ACM

ANM

Voice [Analog]

On hook

Voice [G.711]

MDCX

REL

DLCX

RLC

Ring back

CRCX

RLC

On hook

REL

Busy

B

PSTN

IP

IP Network

TGW 1

MGC

TGW 2

Obs.: There are ACK in MGCP messages, not represented in draw