ITU-T Kaleidoscope Conference
Innovations in NGN

Interoperability Problems in Next Generation Network Protocols

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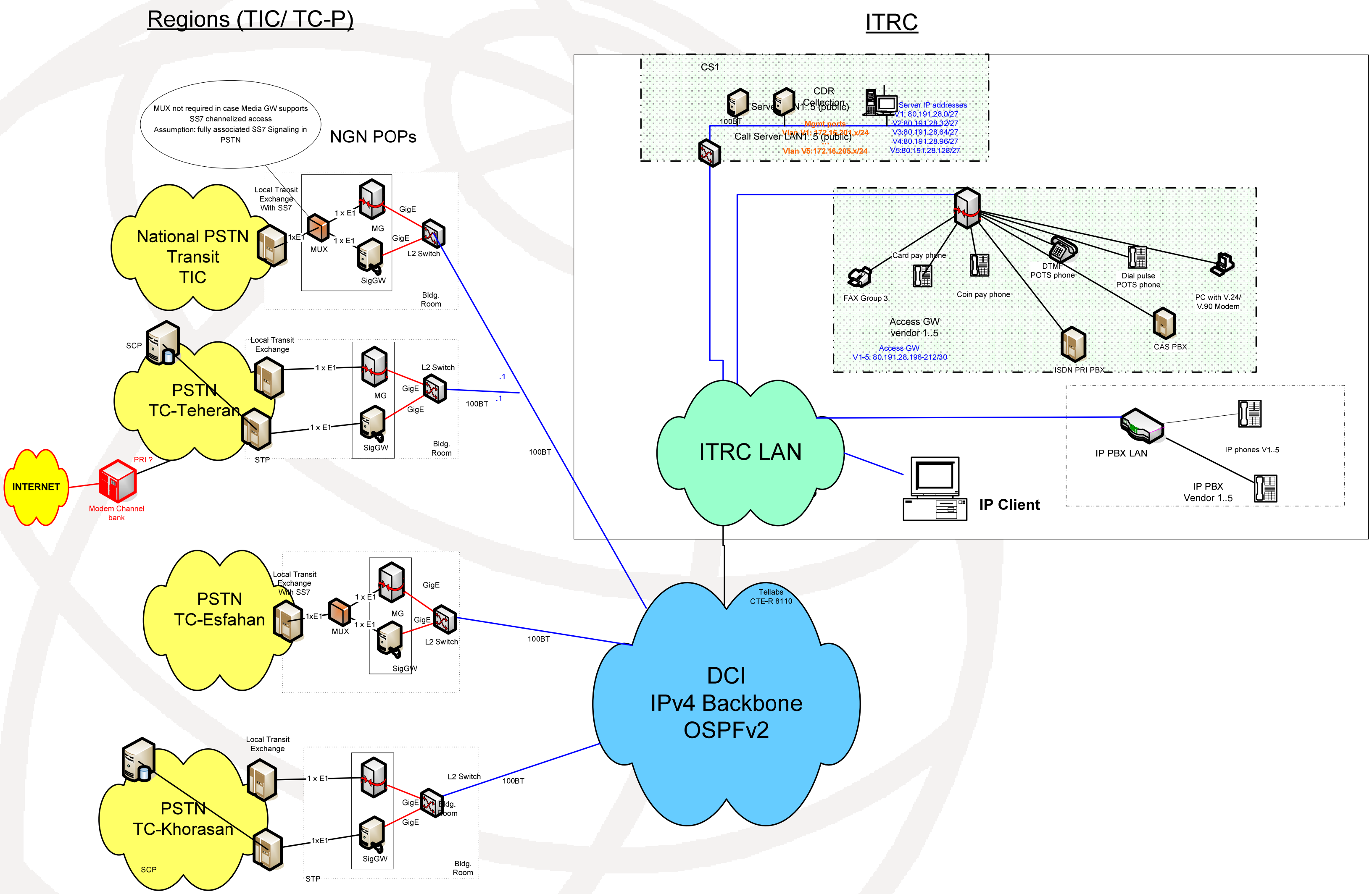
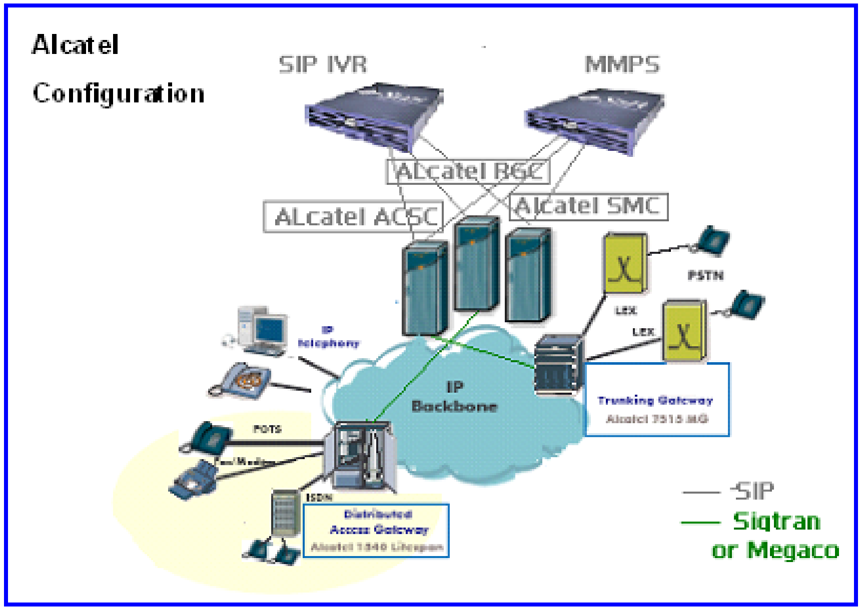
Outline

- Pilot Architecture
- Vendors Solution
- Participated Protocols
- Test Categories
- Results of Pilot Activities
- Test Execution Process
- Analysis result
  - Causes of Unsuccessful Call Server Interoperability
  - Causes of Unsuccessful Gateway Interoperability

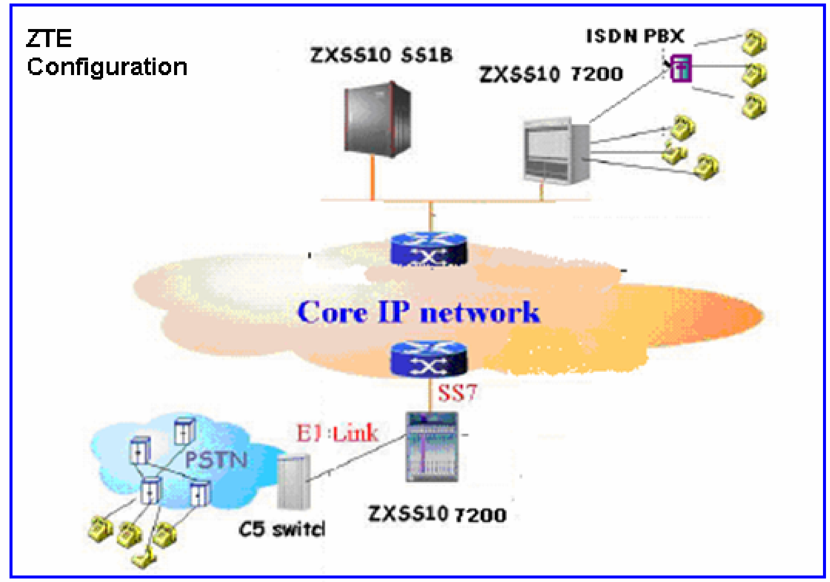
Pilot Architecture



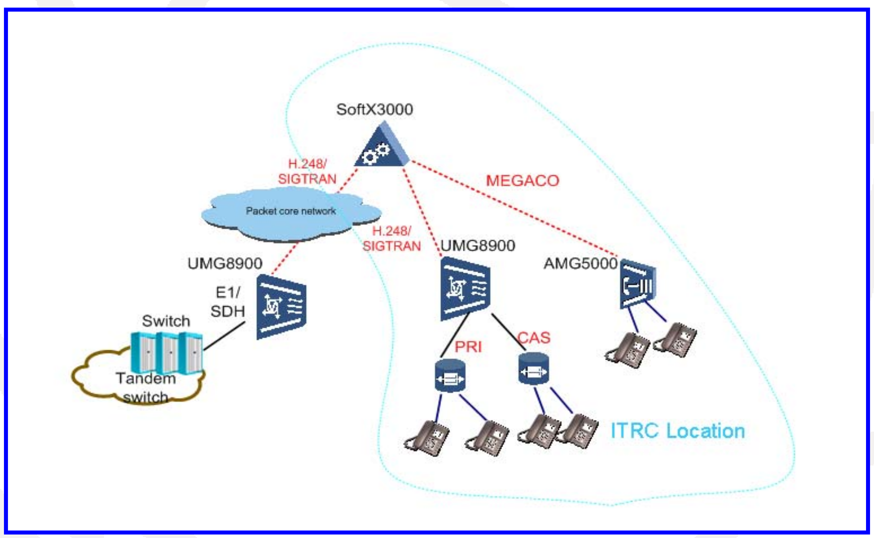
Alcatel NGN solution



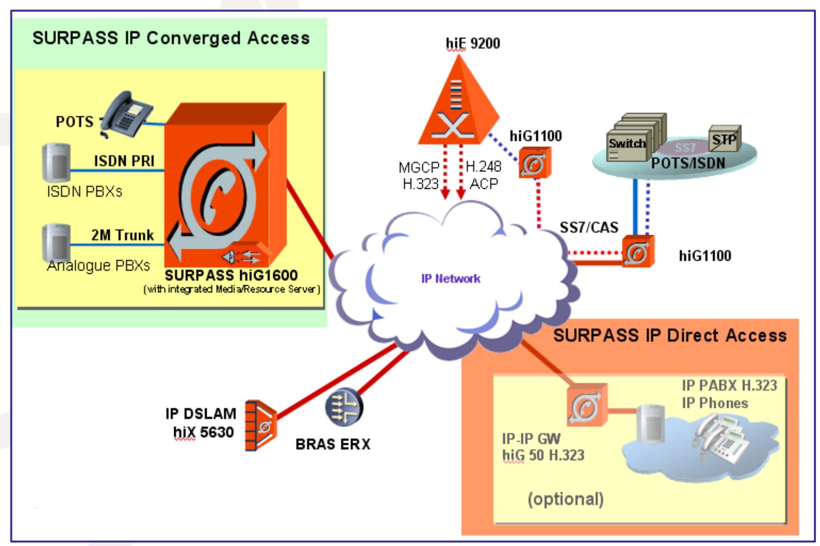
ZTE NGN solution



Huawei NGN solution



Siemens NGN solution



Participated Protocols

IP Signaling Protocol

IP Protocol	Between
Megaco/H.248	CS, AG
	CS , MG
Sigtran/M3UA	CS , SG
Sigtran/IUA	CS , MG
SIP	CS , SIP PBX
	CS , SIP Client
H.323	CS , H.323 PBX
	CS , H.323 Client
SIP-T	CS , CS

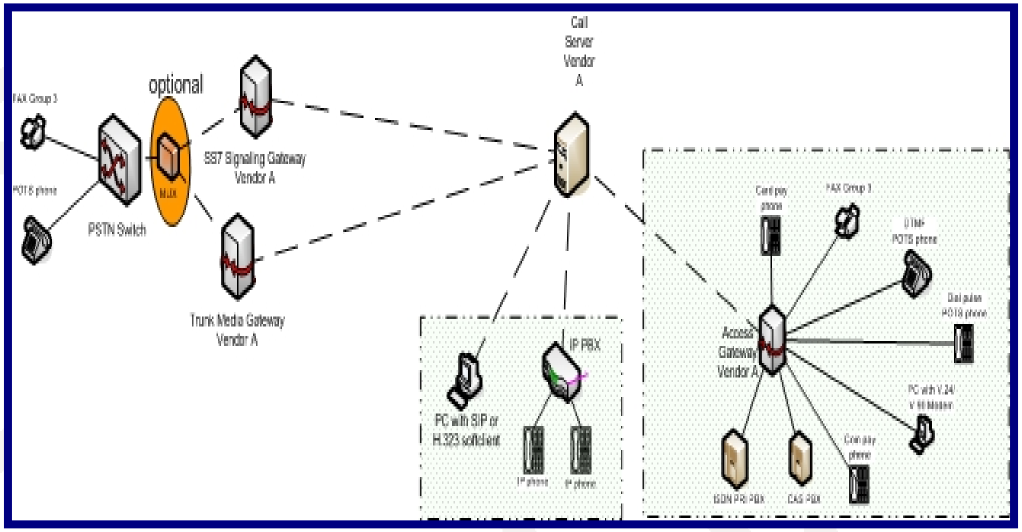
TDM Signaling Protocol

TDM Protocol	Between
SS7/ISUP	SG , PSTN
3bit CAS	MG , CAS PBX
ISDN-PRA	MG , PRA PBX

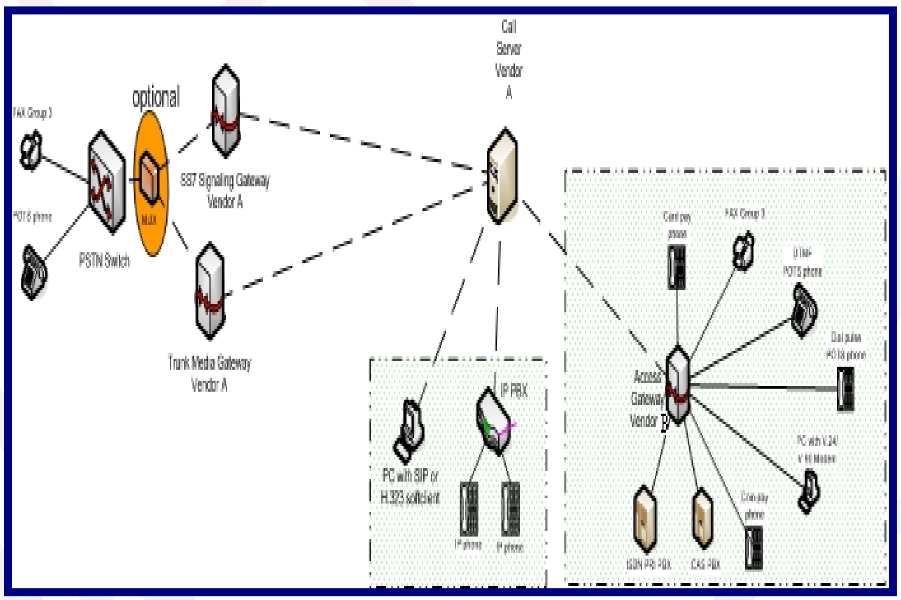
Test Environment

- Single vendor
- Call Server Interoperability
- Gateway Interoperability

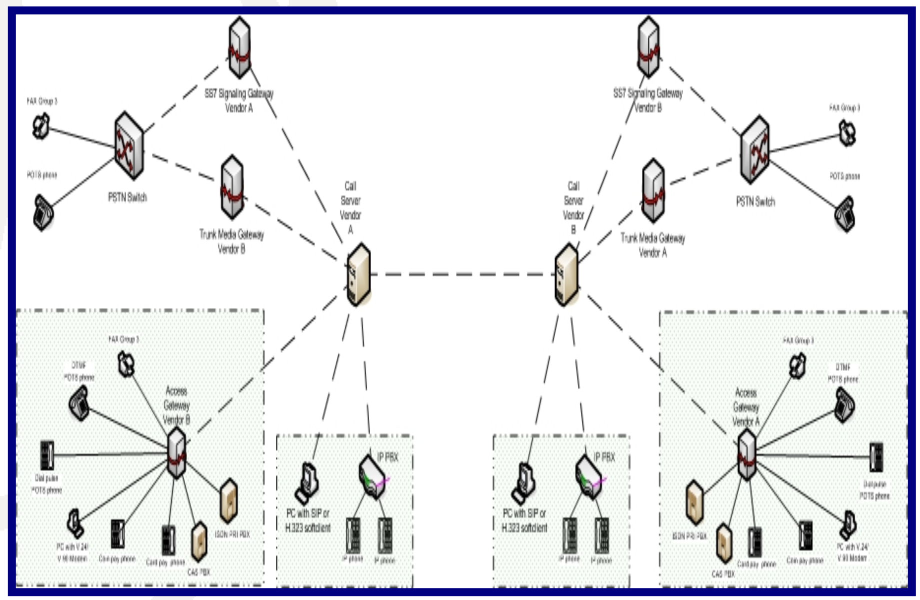
Single Vendor



Gateway Interoperability



Call Server Interoperability



Test Category

1. PSTN values and NGN interface on access gateways
2. Basic Call
3. Residential supplementary services
4. IP Centrex services
5. Regulatory services
6. OAM/NMS (FCAPS) & upgrading

Detail Test Category

Basic call(with G.711/G.729/G.723 codec)	Supplementary services for residential users	IP Centrex services	Regulatory Services	OAM/NMS (FCAPS) & upgrading
Basic Call (Voice)	Abbreviated dialling	Basic call	Emergency call routing	CDR Basic
Basic Call (Busy B-Party)	Outgoing call barring	Private numbering plan	Malicious call identification (MCID)	CFU CDR
Basic Call (Invalid B-Party)	Do-not-disturb	Calling line identification presentation	Legal intercept	LCDR
Basic Modem	Malicious call identification	Direct Call pickup		Configuring services on Graphic User Interface
Basic Fax	Call forwarding (unconditional/busy/No reply)	Group Call pickup		Fundamental Configuration Management capabilities of the Pilot configuration
Basic Call with CC	Call completion	CallPark & CallPickup		Defining & Configuring the components of NGN Network
	Call transfer	Do not disturb		Performance Management capabilities
	Call Hold	Do Not Disturb & Personal Voice Mail		Configuring IP Trunk
	Call waiting	Outgoing Call Barring		Changing of Codec
	Three-party-service	Simultaneous Ringing		CDR Management capabilities
	Meet-me Conference	Ringer Times		Hardware upgrade procedure (HW Upgrade)
	Operator position with Barge-in	Click to Call and Call logs		Software upgrade procedure (SW Upgrade)
	Calling line identification presentation	Hunting Groups		Software patching procedure (SW patching)
	Calling line identification restriction	ComOffice(outlook) – Click to Call		
	Connected Line Identification presentation (COLP)			

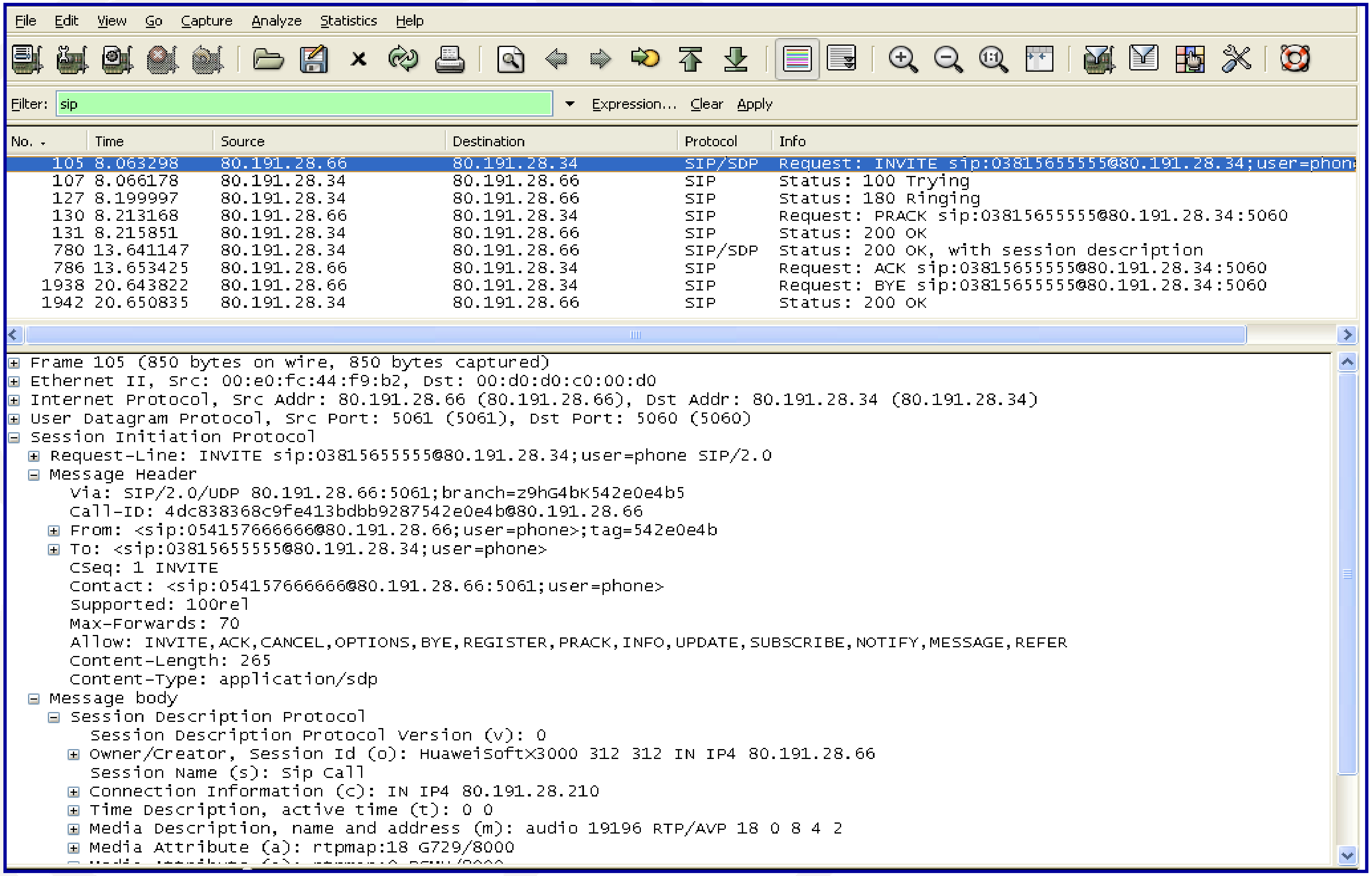
RESULTS OF PILOT ACTIVITIES

- Single vendor
  - Fail of few modem connection using G.729
- Call Server Interoperability
  - Fail of some basic call tests in one direction
  - Fail of more than 50% fax connection and about 90% dial-up modem connection using G.729
- Gateway Interoperability
  - Fail of all CS-MG/SG and most CS–AG connection of different Vendors

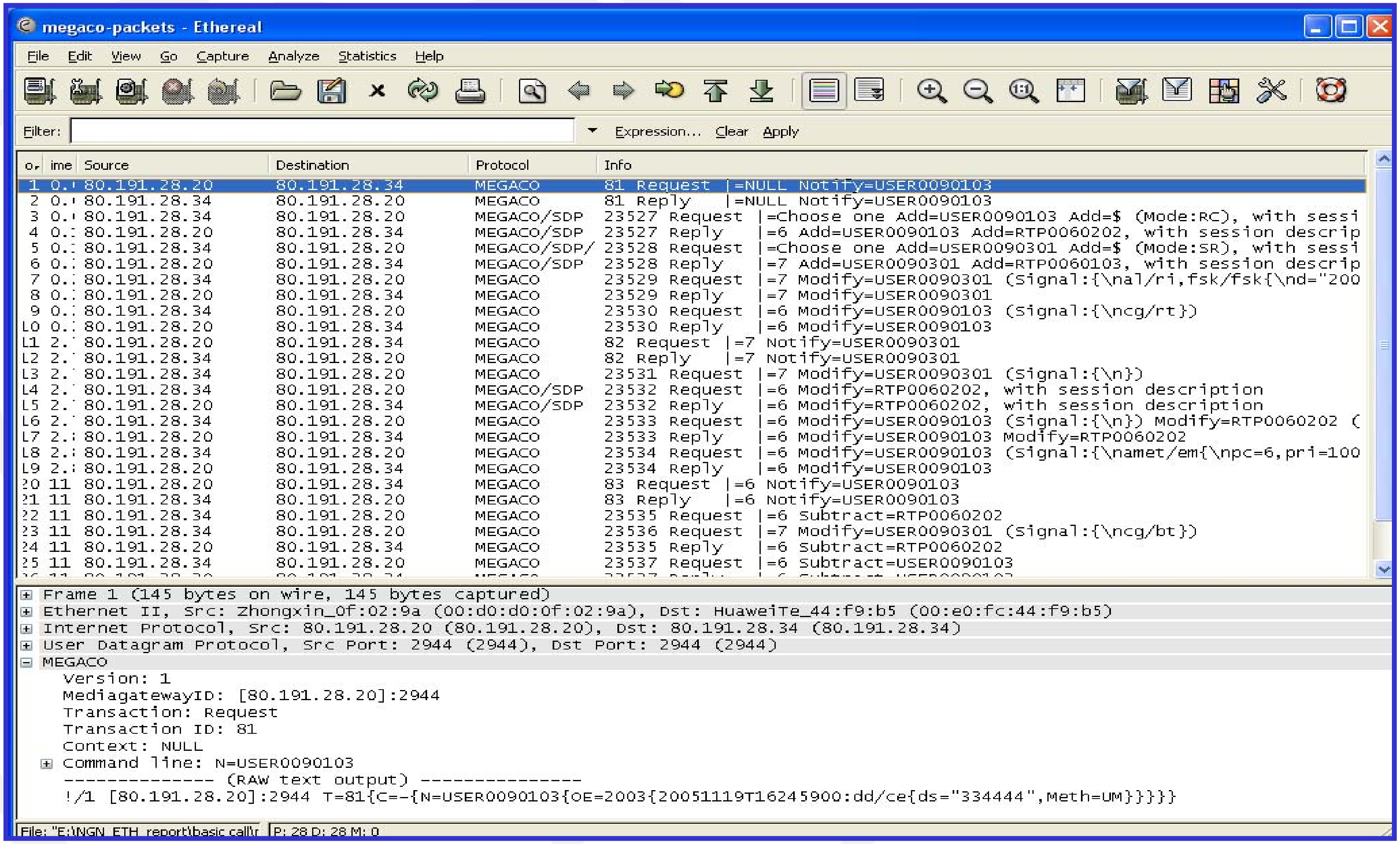
Test Execution Process

- Use of a protocol analyzer and real time sniffing software
- Save captured files and analyze them bit by bit
- Report detail result to the relative vendors

Filtered Messages for SIP Protocol (an example)



Filtered Messages for Megaco Protocol (an example)



Analysis Result

- Causes of unsuccessful Call Server Interoperability (in SIP Protocol)
- Causes of unsuccessful Gateway Interoperability (in Megaco Protocol)

Causes for Unsuccessful Call Server Interoperability

- Privacy mechanism didn't support by call server and it responded by a 420 (bad ext)
- Reinvite message didn't send in Fax or Modem calls using G.729 to change coding protocol into G.711.
- Non standard Header: Non standard P-Asserted-Identity header in the invite message disconnected call by 400 (bad req). Non standard Warning header also disconnected call by 488 (not acceptable header)

Causes for Unsuccessful Call Server Interoperability (*cont.*)

- Packet Length: The call server which was unable to receive messages more than 1300 bytes on UDP transport disconnected call and responded by 513 (message too large).
- Proxy Route Processing
- 5xx Responses which are due to Servers Internal Error.
- Timeout response (408) was not sent because Response timer was not set.

Causes for Unsuccessful Call Server Interoperability (*cont.*)

- Dual Release Issue: one side of the connection repeats BYE message for seven times and caused a lot of signalling traffic.
- PRACK Service which is needed to support reliable transmission did not supported. So destination was not received it, timeout occurred and session was disconnected by sending CANCEL message.

Causes for Unsuccessful Call Server Interoperability (*cont.*)

- Unable to Flash and assumed it as an onhook message so some of supplementary service tests failed.
- Different mechanism in supplementary services for example call completion to busy subscriber service. Some vendors not supported that in SIP-T interface and respond with 488(not
- accepted here). One of them sent invite message periodically for destination when the service activated

Causes for Unsuccessful Gateway Interoperability

- Only in gateway registration phase "digit map" package has been sent to it while the gateway has waiting for each call
- Using different package for pulse detection (MFD and DD)
- Using Fixed Termination ID name for access gateway in the call server.
- In a modem – modem connection Xcg/spec message sending by one side can not been detected by the other

Protocol References

- RFC 3015: Megaco Protocol Version 1.0
- RFC3261: Session Initiation Protocol
- RFC3262: Reliability of Provisional Responses in the Session Initiation Protocol (SIP)
- RFC3324: Short Term Requirements for Network Asserted Identity
- RFC3325: Private Extensions to the SIP for Asserted Identity within Trusted Networks

Protocol References (cont.)

- RFC3323: A Privacy Mechanism for the Session Initiation Protocol (SIP)
- RFC3332 : SS7 Message Transfer Part 3 (MTP3) - User Adaptation Layer (M3UA)
- RFC3057: ISDN Q.921-User Adaptation Layer
- Q1912: Interworking between SIP and Bearer Independent Call Control Protocol or ISDN User Part