TATA COMMUNICATIONS

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Tata-TELUS-Telekom Austria IPX Trial Report

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1.0	**09/29/08**	Robert Bobkoskie	

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1 Introduction

This report summarizes the findings from an IPX trial that included Tata Communications, TELUS and Telekom Austria participating as carriers. TELUS, along with Hutchison 3G (H3G) participated as MNOs. The equipment used in the IPX included: Sonus Networks, Acme Packet and Nortel.

2 Test Objectives

The goal of the testing was to demonstrate and verify packet voice inter-working and cascade billing in the IPX. **Table 1** summarizes the test objectives of this trial.

Features	Description
Protocol Specification	ITU-T Q.1912.5, ANSI88, ITU-T92+
Codec	G.711, G.729
Call Establishment	A and B party establishment. Ring tone generation Abnormal conditions, clean handling of calling busy subscriber, clearing before completion, etc
Call Clearing	Normal cases A and B. Abnormal cases – confirm clean handling with appropriate user indications
Supplementary Services	CLIP; CLIR
Packetisation Period	20ms
DTMF	In-band transport (G.711), Out of band (RFC 2833, G.729)
Echo Cancellation	None.
Cascade Billing, see **Section 2.1** for further detail	CDR Generation to determine duration of open media channel. Collection of expected event labels and cause codes
QoS, see **Section 2.2** for further detail.	Subjective verification

Table 1: Summary of Testing

2.1 Cascade Billing Scope

The goal of the testing was to demonstrate and verify that the appropriate billing information was found in CDRs in each part of the end-to-end chain. This was restricted to a simple voice service for the tests. All other services were out of scope.

In general the focus for cascade billing was the verification of:

- CDR production within the technical infrastructure
- The content of the CDR
- The CDR content requirements related to the specific scenarios
- The manual identification, exchange and comparison of the generated CDRs out of all platforms related to the same test calls.

Where feasible the following scenarios were to be trialled:

- National and international interconnect
- Calls from and to MNOs and FNOs
- Calls across different vendor platforms
- Very short duration calls
- Long duration calls
- The handling of ineffective (i.e. non-chargeable) calls
- Calls from one time zone to another

2.2 QoS Testing Scope

When voice is carried over IP it needs to offer a quality of service equivalent to that experienced by using circuit switched technology between the same two destinations. Voice QOS was subjectively verified by test during the tests and providing a qualitative assessment of the perceived call quality.

2.3 Trial Specification, Test Plan and Cases

The test cases executed for this trial were obtained from [1].

2.4 Test Configuration

Figure 1 displays the configuration used in this trial. Tata Communications was configured to preserve call control information by encapsulating ISUP, utilizing SIP-I (Q.1912.5) as follows:

- ANSI between Tata Communications and TELUS
- ITU-T between Tata Communications and Telekom Austria

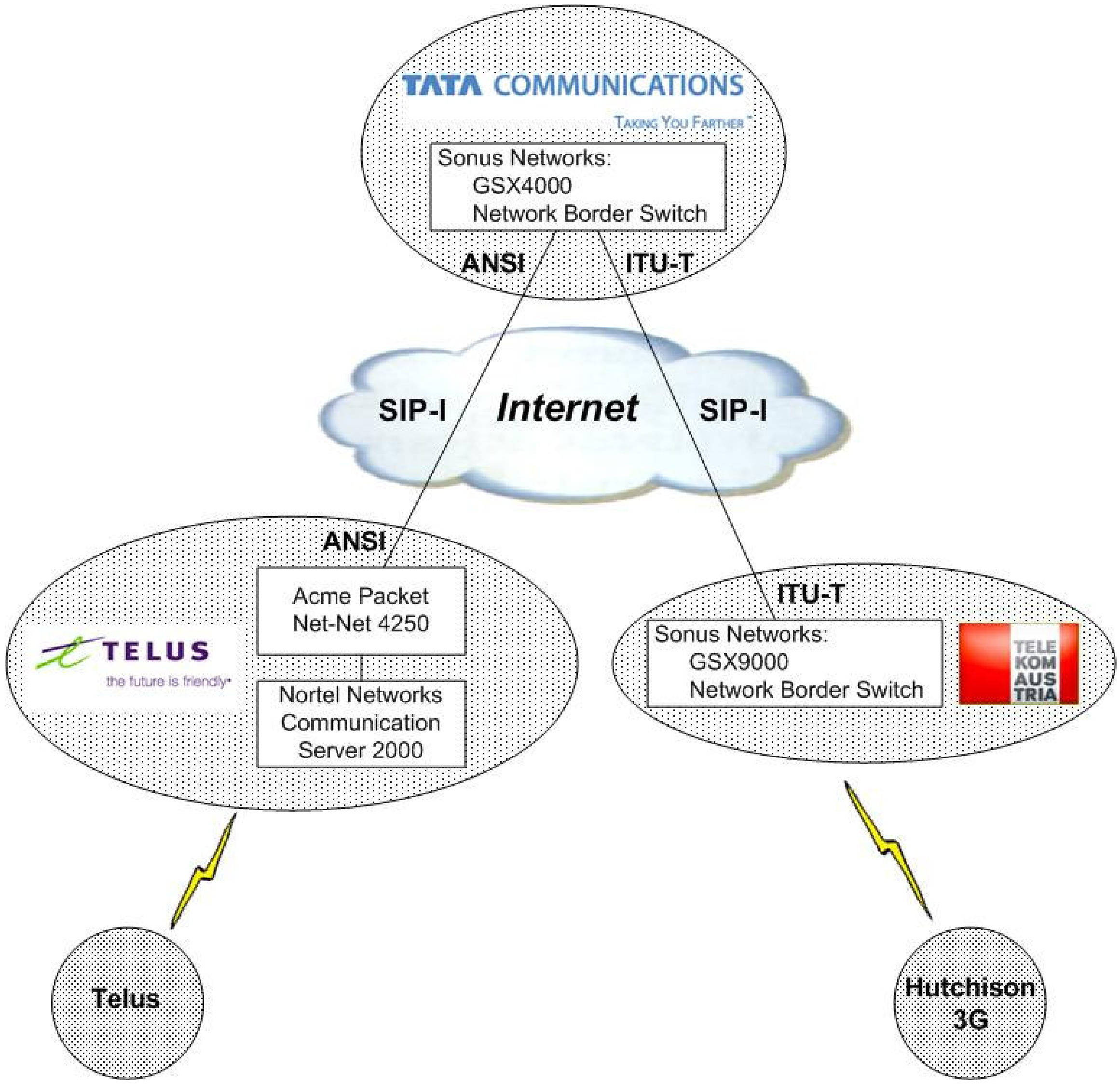


Figure 1: IPX Configuration

2.5 Equipment

The following equipment was utilized for this trial.

Equipment	Software / Firmware Version
Tata Communications • Sonus GSX4000	V07.01.00.A701
TELUS • Acme Packet Net-Net 4250 • Nortel Networks Communication Server 2000	C6.0.0 Patch 3 (Build 60) SN09
Telekom Austria • Sonus GSX9000	V654.R001

Table 2: Equipment List

3 Conclusion

This trial successfully verified packet voice inter-working and cascade billing in the IPX. More specifically, voice traffic was exchanged between MNOs via the IPX, and included:

- Verify packet voice inter-working (PVI) using SIP-I connectivity
- Validate interoperability between two distinct ISUP standards:
  - The ANSI standard, ANSI88
  - The ITU-T standard, ITU-T92+

3.1 Observations and Issues Found

The following issues were found during this IPX trial. Note that issues regarding interoperability are broken into the following bins:

- General scheduling issues due to differences in time zones
- Interoperability between ANSI and ITU-T 'Base' Variants
- Interoperability between Partners' Network Implementation
- Interoperability with SIP, ISUP and E.164
- IP Fragmentation

3.1.1 General Scheduling Issues

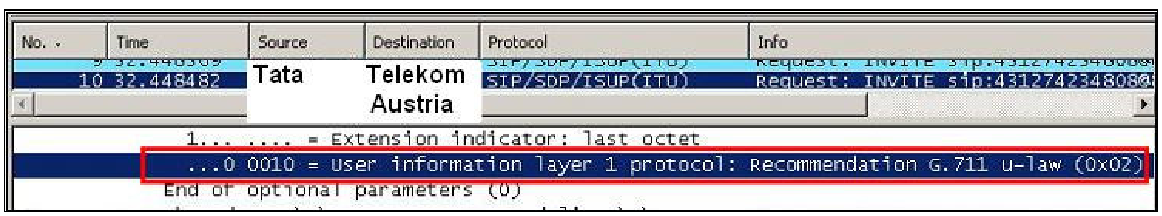
Due to the differences in geographic location and corresponding time zones for the participants in this trial, scheduling became an issue.

3.1.2 Interoperability between ANSI and ITU-T 'Base' Variants

Due to differences in the parameters defined by the ANSI and ITU-T 'Base' variants, many issues regarding interoperability required resolution. The trace below shows that ISUP was not encapsulated (via SIP-I, Q.1912.5) between Tata and Telekom Austria due to (ANSI88) parameters received from TELUS that were not conformant with the ITU-T92+ variant. Resolving this issue required modifying or eliminating parameters on either the ANSI88 or ITU-T92+ variant.

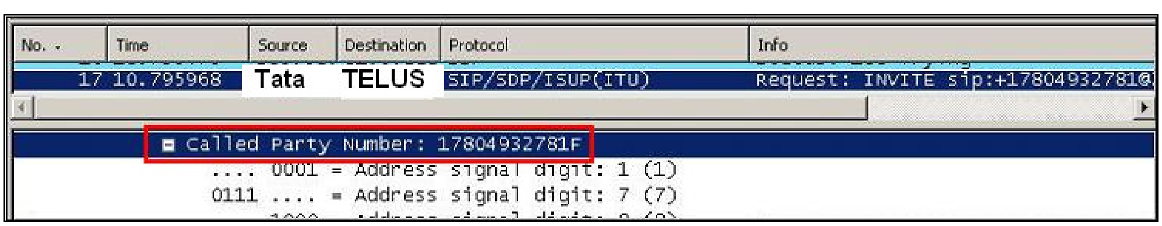
3.1.3 Interoperability between Partners' Network Implementation (1)

Successfully completing a call across the IPX network required the resolution of issues regarding our partners' network implementation. These issues corresponded to the network elements used by our partners, as well as unique parameters/settings required/utilized for their respective VoIP/TDM networks. The trace below shows that G.711 u-law was included in the encapsulated ISUP sent from Tata to Telekom Austria. Since Telekom Austria uses G.711 a-law, the call to failed on the TDM part of Telekom Austria's network. Resolving this issue required eliminating the User Service Information sent to Telekom Austria.



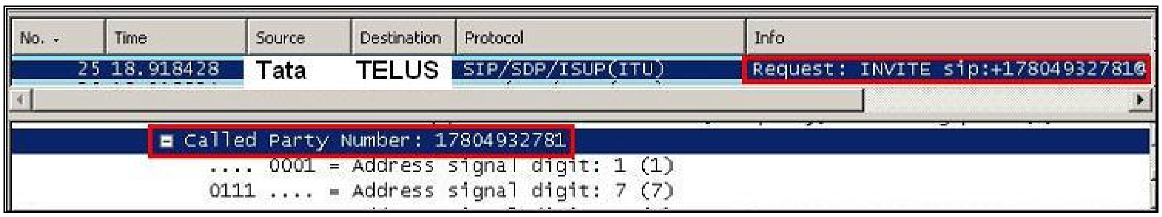
3.1.4 Interoperability between Partners' Network Implementation (2)

Another example regarding interoperability between our partners' network implementation involved the use of a 'stop' bit to terminate the received digit string for the Called Party Number (CPN). The trace below shows that a 'stop' or 'F' bit was sent in the ISUP CPN to TELUS, which eventually caused the call to fail on the TDM part of the TELUS network. Resolving this issue required eliminating the 'F' bit in the CPN sent to TELUS.



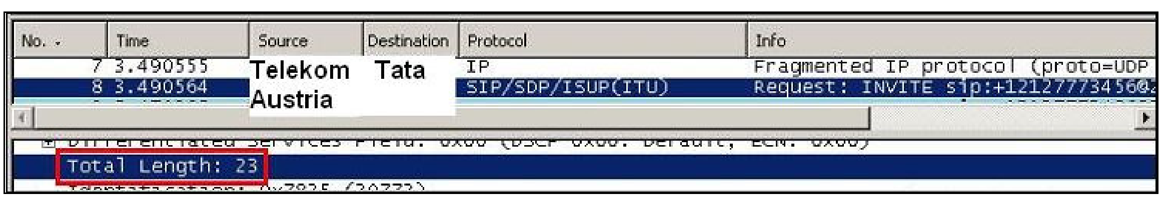
3.1.5 Interoperability with SIP, ISUP and E.164

Due to requirements for call routing in our partners' VoIP networks, SIP INVITEs were formatted according to the E.164 standard. There was an issue where the leading '+' symbol, utilized in the E.164 standard, was required in the SIP 'URI' and 'To' field to facilitate VoIP call connectivity. However, the '+' symbol caused issues with ISUP encapsulation, therefore a solution to include the '+' symbol for SIP and exclude it for ISUP was required. The trace below shows an example of the format required to complete calls to TELUS VoIP and TDM networks. Note that the issue regarding the 'stop' bit shown in the previous slide has been addressed.



3.1.6 IP Fragmentation

Due to ISUP encapsulation, SIP messages are larger and IP fragmentation may occur. There were scenarios where fragmentation allowed for IP packets with small payloads, which were consistently dropped over the IP network. This caused issues with B2BUAs waiting for additional fragmented packets and thus not having the ability to reconstruct SIP INVITE messages if these packets were dropped. The trace below, obtained from Telekom Austria's network, shows a packet originating from Telekom Austria. The packet, part of a SIP INVITE message was fragmented with one packet having a total length of 23 bytes. This packet was dropped in the IP network, never arriving at the terminating side. Resolving this issue required eliminating codecs that were sent in the SDP from Telekom Austria. Doing this allowed for 'shorter' SIP messages that were not fragmented. For this trial, SIP was transported via UDP. It may have been possible that using TCP might have addressed this issue, however this was not verified.



3.2 Helpful Hints for Future Trials

The following procedures were found to be helpful in addressing the issues found in **Section 3.1**.

- Use of 'loopback' numbers at the edge of the IPX network. This allowed for participants to verify call origination/termination independently of other trial participants. This addressed the differences in time zones between partners.
- Having engineers with experience in both SIP and ISUP was important for debugging end-to-end call flows. This addressed the issues regarding interoperability and IP fragmentation.

4 Appendix A: Abbreviations

Term	Definitions
B2BUA	Back to Back User Agent
CDR	Call Detail Record
CFNR	Call Forward on Not Reachable
CFU	Call Forward Unconditional
CLIP	Calling Line Identification Presentation
CLIR	Calling Line Identification Restriction
CW	Call Waiting
DTMF	Dual Tone Multiple Frequency
EFR	Enhanced Full Rate
FNO	Fixed Network Operator
GSMA	GSM (Groupe Spéciale Mobile) Association
IETF	Internet Engineering Taskforce
IP	Internet Protocol
IPX	Internet Packet Exchange
ISUP	Integrated Services Digital Network User Part
ITU	International Telecommunications Union
MNO	Mobile Network Operator
PVI	Packet Voice Inter-working
QoS	Quality of Service
RFC	Request for Comments
SBC	Session Border Controller
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SIP-I	SIP with encapsulated ISUP
SP	Service Provider
UI	User Interface
VoIP	Voice over Internet Protocol

5 Appendix B: References

1. GSMA IPX PCI SIP-I Packet Voice Interworking v1.4, June 12 2008.