3rd Annual i3Forum Conference

The Future is All IP

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Chicago

Operations Perspective of VoIP Migration

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Agenda

Scope of activities

Migration status – how are we doing?

Operations WS deliverables

Planned QOS trial

Scope of activities

- Provide tools, processes and guidance to facilitate VoIP migrations
- Provide day to day perspective of VoIP challenges and issues related to provisioning, integration and operational support
- Tracking and trending of VoIP migrations

VoIP Migration – last 3 year evolution

- 2 to 3 years ago there were significant challenges for carriers
  - Interop between vendors
  - SIP-I issues
  - Fax issues
  - Training was needed
  - Internal processes and procedures were needed for VoIP
  - NGN migrations were just starting

Where are we today?

- Good news is that most significant technical issues are now behind us!
  - Focus now needs to be on continued execution of new VoIP interconnects
  - Completion of migrations and removal of legacy TDM
  - Networks and technology is now in place to allow integration of new services and capabilities
  - Move forward with QOS solutions

VoIP migration data – i3 forum internal survey

- 11 companies provided data sample
  - Inbound, outbound and end to end data
  - March 2011 to March 2012
- Highlights:
  - Inbound %increased by 22%
  - Outbound %increased by 11.5%
  - End to end %increased by 15%

Summary	Mar-11	Apr-11	May-11	Jun-11	Jul-11	Aug-11	Sep-11	Oct-11	Nov-11	Dec-11	Jan-12	Feb-12	Mar-12	% change
Inbound Average	24.21%	25.18%	25.26%	26.45%	26.71%	27.44%	28.10%	28.92%	29.63%	30.20%	29.43%	30.12%	29.54%	**22.03%**
Outbound Average	20.46%	20.85%	21.69%	21.89%	20.80%	21.27%	21.46%	22.51%	22.85%	24.40%	22.65%	23.61%	22.82%	**11.52%**
End to End Average	10.54%	10.32%	10.13%	10.91%	10.98%	11.01%	10.56%	12.54%	12.74%	13.18%	12.30%	12.97%	12.12%	**14.92%**

IP Migration kickoff

- Took place on Wed, May 16 @ ITW
- Created event to allow carriers to meet with sole purpose to plan or complete VoIP migrations
- Initial feedback:
  - *"i3 forum IP migration session allowed me to meet with a key customer and in 5 minutes we were able to solve a major problem that had prevented us from migrating for the last year"*

Deliverables 2012

- Interconnection Form for International Voice Services, release 5
  - Focus on key questions on both IP and voice layer that are critical for successful interconnects

- Interoperability Test Plan for International Voice Services, release 5
  - Test cases designed to validate and ensure the VoIP interface will successfully pass voice traffic

Interconnection form for International Voice Services, release 5

Connectivity Form			
Interconnection Configurations		carrier A name	carrier B name
	IP interconnection mode		
Private/Public interconnection			
	IP interconnection options - Private interconnection		
interconnection options ***Note: please see IP Inteconnect examples tab for further details***	Layer 1		
	Layer 2		
	Layer 3		
PoP location	Country		
	City/ State/ Province		
	Address/Suite/Floor/Room/etc		
IP routing	Routing protocol		
Pingable IP Address (PE Router)	IP address for reachability test (ping)		
Traffic marking	IP Media marking		
	Media value		
	IP Signalling marking		
	Signalling value		
	IP interconnection options - Public interconnection		
Interconnection options	direct link (sharing IP transit and VoIP traffic)		
	Public Internet		
Security options	IPSec VPN		
Additional Comments		carrier A name	carrier B name
	Comments Include any additional comments related to interconnect		

Interconnection form

Service Form			
Service Description		**carrier A name**	**carrier B name**
Class of service			
	Class of service		
Capacity			
Concurrent calls	Max number of concurrent calls A->B		
	Max number of concurrent calls B->A		
	Max number of concurrent calls B<->A		
Calls Per Second dimensioning	expected CPS A->B		
	expected CPS B->A		
Signalling Protocols & Parameters SIP-I (Preferred) and SIP		**carrier A name**	**carrier B name**
Signalling protocol			
Signalling protocol details			
SIP-I	ITU-T Q.1912.5 Annex C Prof. C		
	ISUP Version (recommended ITU-T92)		
	ISUP Base (recommended itu-t92+)		
SIP mandatory RFCs	SIP (RFC 3261)		
	An offer/answer model with SDP (RFC 3264)		
	Privacy header (RFC 3323)		
	P-Asserted-Identity (RFC 3325)		
SIP recommended RFCs	Diversion Header (RFC 5806)		
	Reason header field for SIP (RFC 3326)		
SIP optional RFCs	SIP Session Timers (RFC 4028)		
	SIP Update Method (RFC 3311)		
Other	Other signalling protocol details		
Mapping method			
	Identify the mapping recommendation SIP <> ISUP if used		
Signalling IP address			
Signalling IP	IP address(s) or network range to be used		
	Comments:		

Interconnection form

Transport Parameters			carrier A name	carrier B name
Transport protocol				
UDP	Protocol and source/destination port used for transport layer		source portdest. port	source portdest. port
TCP	Protocol and source/destination port used for transport layer		source portdest. port	source portdest. port
SCTP	Protocol and source/destination port used for transport layer		source portdest. port	source portdest. port
Media parameters			carrier A name	carrier B name
Media IP Address				
Media IP		IP address(s) or network range to be used		
		Comments:		
Voice Codec				
Voice Codec, packetization period, VAD support (Highly recommended: G.711 A-law, G.711 µ-law, G.729, G.729a, G.729b, G.729ab)		priority 1		
		priority 2		
		priority 3		
		priority 4		
		Other Codec(s)		
Transcoding		Is transcoding applied?		
Other Services				
DTMF		RFC 4733 (former 2833)		
		G.711 pass-through		
		INFO Method (RFC 6086)		
Fax		T.38 Fax relay		
		G.711 pass-through		
Numbering Format			carrier A name	carrier B name
Numbering				
Called number		+ CC NDC SN (Preferred)		
		CC NDC SN		
		Other		
Calling number		+ CC NDC SN (Preferred)		
		CC NDC SN		
		Other		
Tech prefix before numbering		A -> B		
		B -> A		

Interoperability Test Plan

	Carrier A	Carrier B
Carrier Name		
Date of Test		
Testing personnel contact		
Testing number(s)		
Test fax number(s)		

Test case number	Description	Pass/Fail	Comments
7.1.1	Normal call release – Calling party clears after answer		
7.1.2	Normal call release – Called party clears after answer		
7.1.3	Normal call release – Calling party release while ringing		
7.1.4	Normal call release – Called party release while ringing		
7.1.5	Normal call setup to Ring No Answer / Timeout		
7.1.6	Normal call setup to Busy Line / Calling Party Release		
7.1.7	Verify Proper handling for No Route To Destination		
7.1.8	Verify Proper handling for Unallocated Number		
7.1.9	Verify proper handling for Insufficient Digits		
7.1.10	Verify "long call" duration		

Interoperability Test Plan

7.1.11	DTMF – Verify digits received for a DTMF transmission		
7.1.12	Calling Party Number - Verify that CLI is properly passed and received in the agreed upon format		
7.1.13	Called Party Number – Verify that the called party number is received in the agreed upon format		
7.1.14	CLI Restriction presentation - CLIR (only if agreed upon by both parties)		
7.1.15	Reachability and keepalive mechanism (SIP Options)		

Test case number	**Description**	**Pass/Fail**	**Comments**
7.2.1	Fax transmission test – no fallback		
7.2.2	Fax transmission test – with fallback		
7.2.3	Verify fax image quality		
7.2.4	Verify long fax transmission (3+ page)		

Test case number	**Description**	**Pass/Fail**	**Comments**
7.3	Verify CDR match • Calling number • Called number • Start date/time • Stop date/time • Call duration		

QOS Trial – In planning stage

- Compare the reliability of RTP measurements with probe against RTCP measurements
- Verify the quality/accuracy of the collected data
- Trial to document the implementation effort and deployment experience of passive/active monitoring solutions
- Provide practical experience/application and results to i3 forum/industry based on actual application and usage
- Goal is to partner with multiple vendors to ensure various solutions and implementations are used