



IPNext NGN IP-PBX

High-performance Next Generation IP-PBX Solution

Korea Telecom IP-PBX BMT



AddPac

AddPac Technology

Sales and Marketing

www.addpac.com

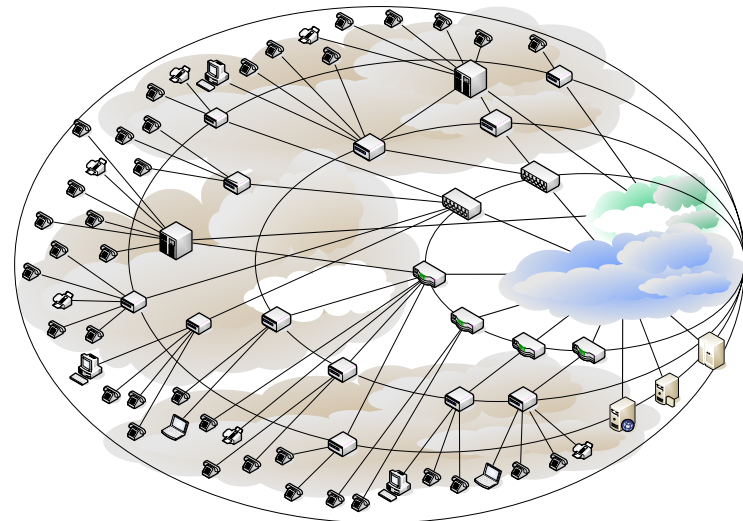
Contents

- IP-PBX BMT Network Diagram
- IP-PBX Performance Testing Environment
- Test Item & Test Result

AP-VP300 Video Phone



AP-IP200 IP Phone

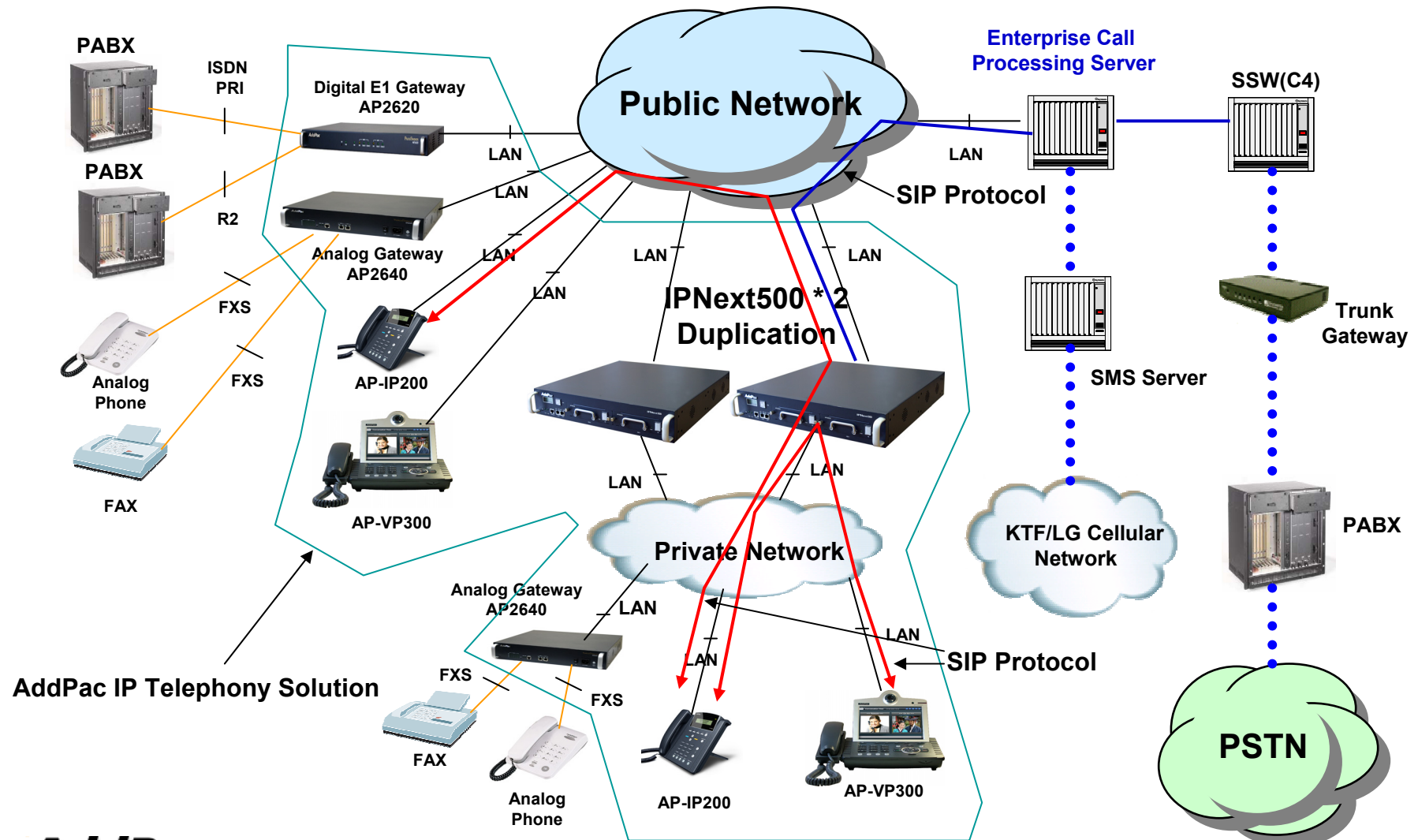


IPNext 200 IP-PBX

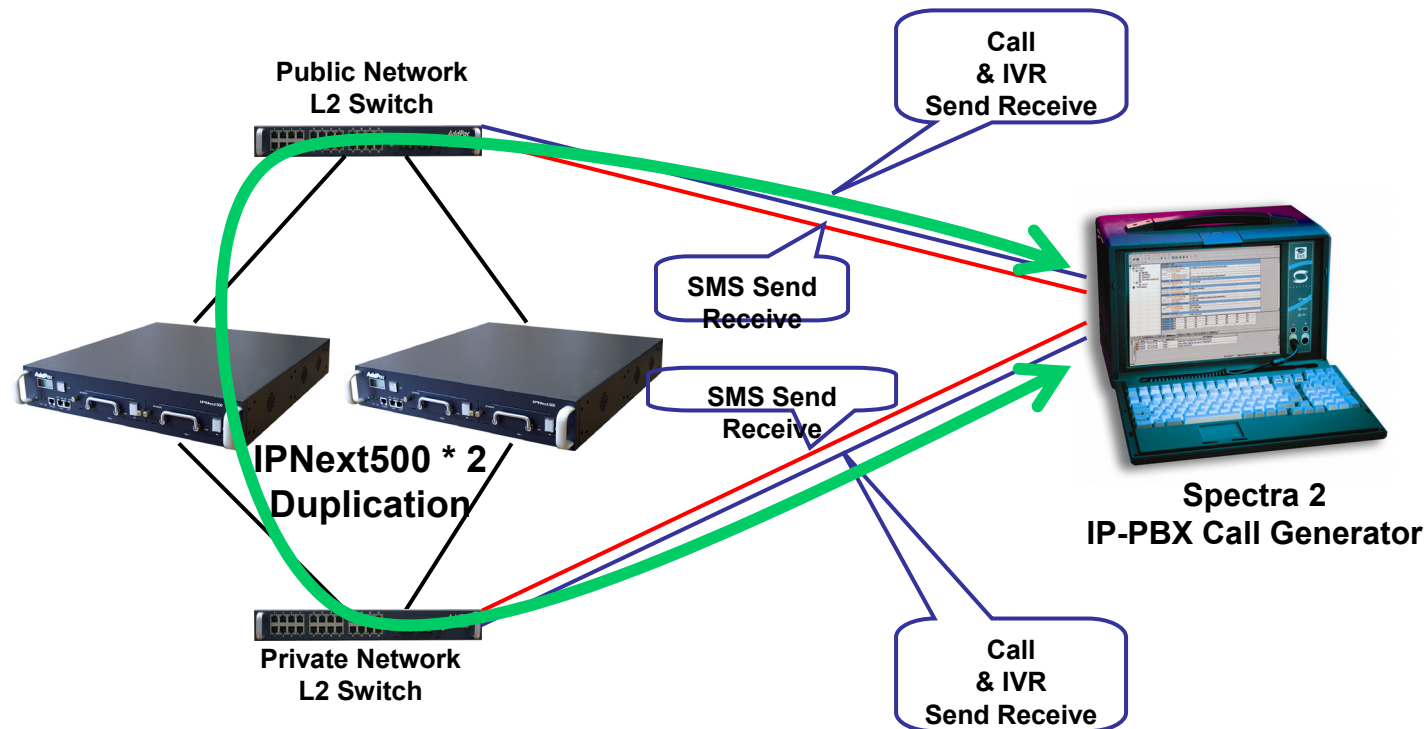


AP-MC1000
Audio/Video MCU

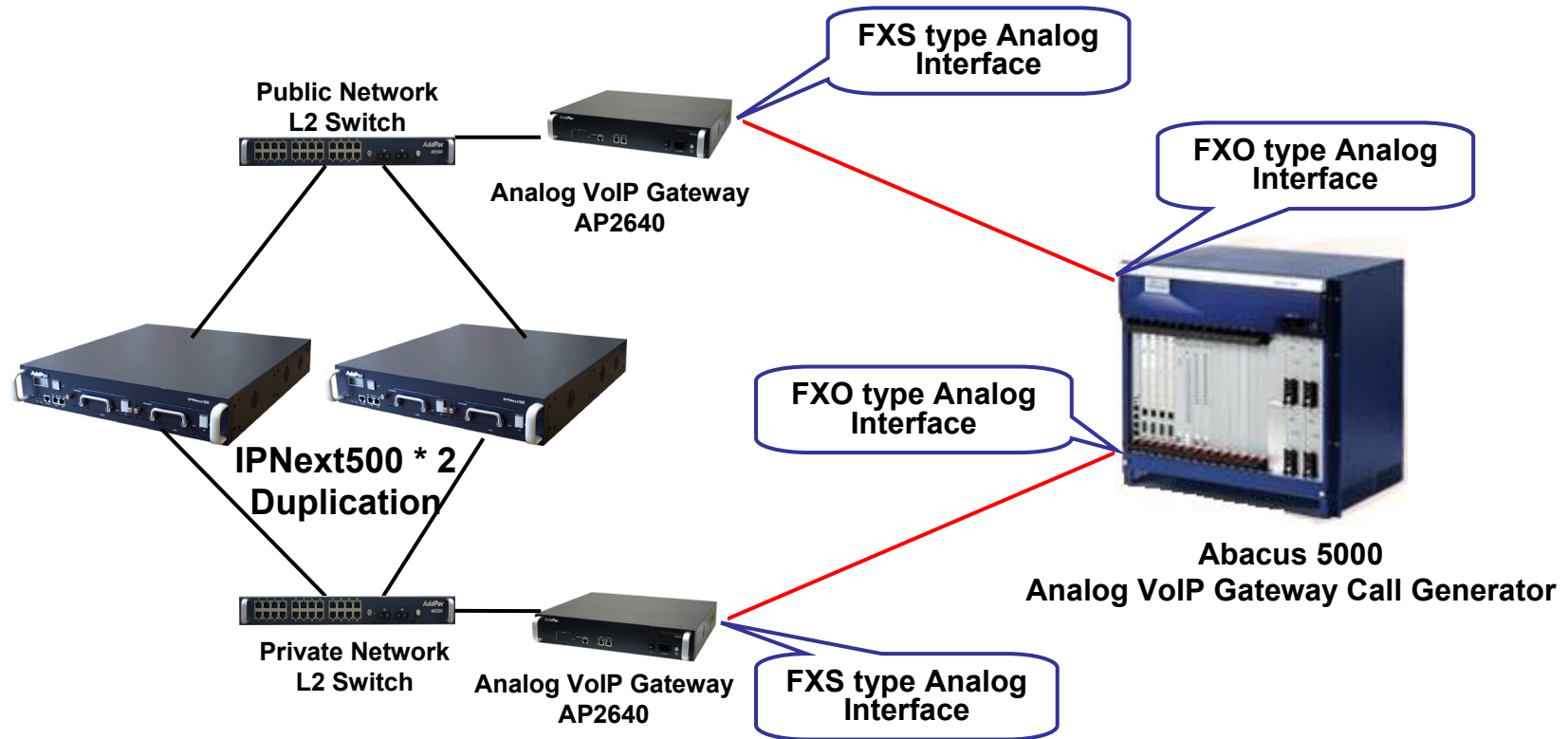
IP-PBX BMT Network Diagram



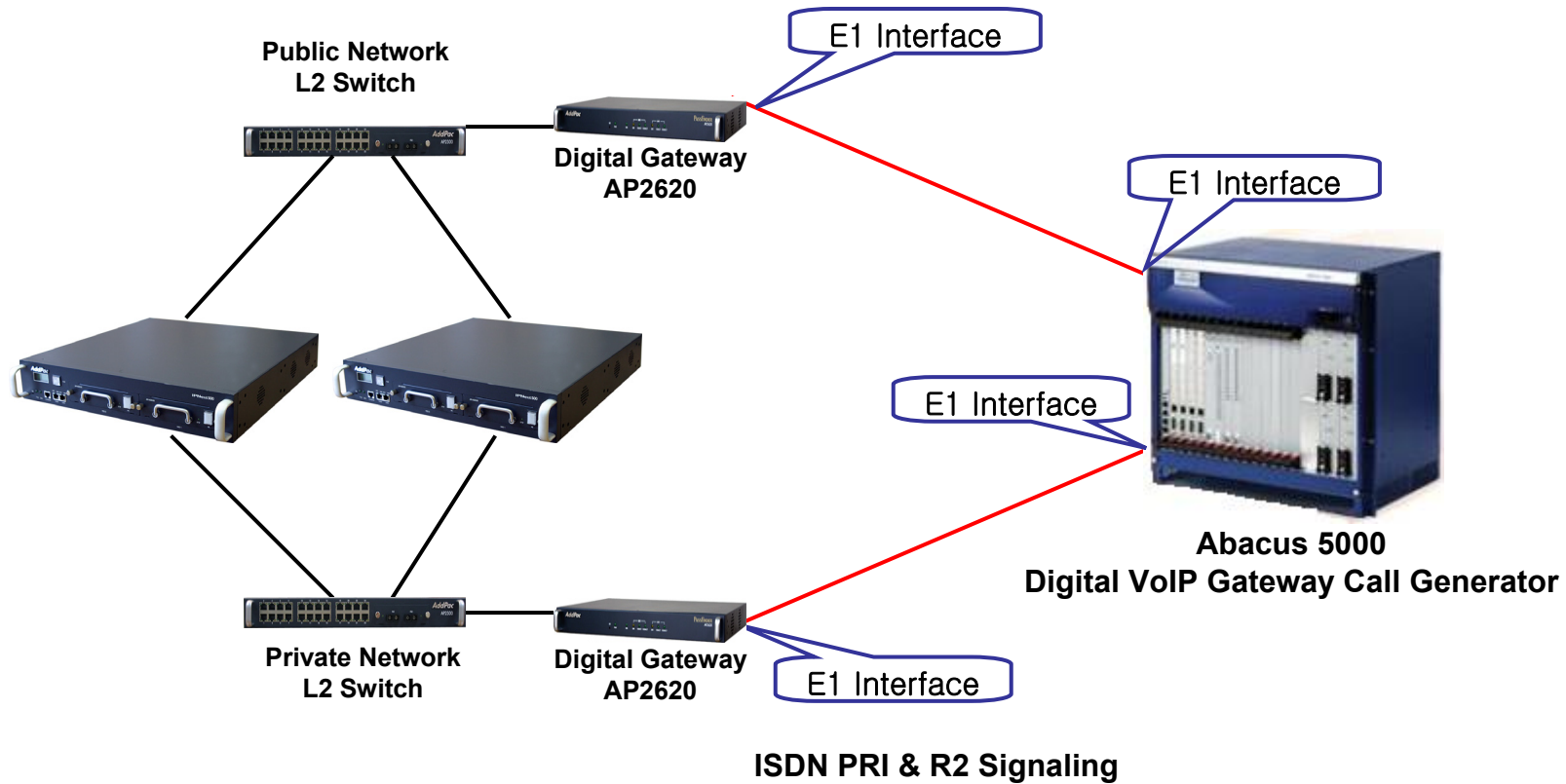
IP-PBX Performance Testing Environment



IP-PBX+ Analog VoIP Gateway Testing Environment



IP-PBX+ Digital VoIP Gateway Testing Environment



IPNext IP-PBX Test Result

System Basic Requirement

Classification	Remark	Result
Basic Function	KT propriety Inter-working specification support for KT Softswitch	○
	IPv4, IPv6 (Optional) Protocol Stack	○
Subscriber Information	KT Terminal Number, Domain, Authentication Password Setting & Configuration Support	○
	Terminal(IP Phone, Video Phone, etc) Number Support for extension line	○
System Architecture	Incoming call receiving method - Direct Incoming Call Receiving using Personal Terminal Number - Indirect Incoming Call Receiving using IVR and Representation Number	○
	Public/Private IP Phone Support	○
	Inter-working Support between IP-PBX and KT Softswitch	○
	IP-PBX Redundancy Support (Active-Active or Active-Standby)	○

IPNext IP-PBX Test Result

SIP Protocols

Classification	Remark	Result
SIP Message (Register)	Support RFC 3261(Session Initiation Protocol) and RFC 3665(SIP Basic Call Flow example)	○
	Registration Support using Representation Number, Domain, Password Setting	○
	Secure message transmission using RFC 2617(HTTP Authentication: Basic and Digest Access Authentication) protocol for registration information protect	○
	Representation Number Registration only at site registration.	○
	Re-Registration periodically depend on expire field value.	○

IPNext IP-PBX Test Result

SIP Protocols

Classification	Remark	Result
SIP Message (Invite)	SIP INVITE message generation and processing at inbound & outbound call	○
	Outbound SIP INVITE message inter-working between IP-PBX and KT Softswitch	○
	Inbound SIP INVITE message inter-working between IP-PBX and KT Softswitch	○
	Support direct call receiving at Inbound INVITE message processing	○
	In case of No Response Message from KT Softswitch at SIP INVITE message transmission - SIP INVITE message re-transmission according to RFC3261 scenario - Call clear in case of No Response continuously	○
	CID transmission support using P-Asserted-Identity (RFC3325)	○
	INVITE message without SDP info. processing	○
	INVITE message from unknown Softswitch, response blocking	○

IPNext IP-PBX Test Result

SIP Protocols

Classification	Remark	Result
SIP Message (Cancel)	In case of No Response Message from KT Softswitch at SIP CANCEL message transmission - SIP CANCEL message re-transmission according to RFC3261 scenario - Call clear in case of No Response continuously	○
SIP Message (Bye)	In case of No Response Message from KT Softswitch at SIP BYE message transmission - SIP BYE message re-transmission according to RFC3261 scenario - Call clear in case of No Response continuously	○
SIP Message (Recode-route)	At Message Receiving including Record-route Header - Add this Record-route information on Route Header of next generation message - Send this message to Server(server address in record-route)	○

IPNext IP-PBX Test Result

IP terminal inter-working

Classification	Remark	Result
IP-PBX Terminal Inter-working	IP Terminal - IP Phone : AP-IP200 - Video Phone : AP-VP300	○
	VoIP Gateway - Analog VoIP Gateway : AP2640 (32Port FXS, FXO, E&M) - Digital VoIP Gateway : AP2620 (Up to 2E1, 2T1)	○
	Support the IP Terminal and Analog Terminal (Phone, FAX) both	○
	Support the all supplementary service (IP terminal and Analog Terminal both)	○
	Inter-working between IP-PBX terminal and pre-deployed KT terminal	○
	Check the Inter-operability between IP-PBX terminal and coming new KT terminal	○

IPNext IP-PBX Test Result

VoIP Gateway inter-working and Dial Plan

Classification	Remark	Result
PABX and LE Inter-working	Interworking with PABX or LE via Digital E1 Interface (External G/W)	○
	Support MFC R2 and ISDN-PRI	○
	CID Number Transmission at LE interworking	○
Routing and Dial plan	Support IP-PBX call between users via Internal phone number dialing	○
	Call Routing to KT Softswitch at outbound call	○
	Support Number addition, Deletion Function about inbound and outbound call	○
	Support Call Restriction about a specific line number or Prefix	○

IPNext IP-PBX Test Result

Media Processing

Classification	Remark	Result
Media Processing	VoIP Codec -IP Terminal and VoIP Gateway : G.711(A/u-Law) , G.729a -VMS (Voice Mail Server) and IVR : G.711(A/u-Law)	
	RTP packet interval -G.711 : 10ms and 20ms -G.729 : 20ms and 30ms	○
	Support Codec Auto Negotiation	○
	Codec Setting and Configuration Management - Codec Priority Setting - Single Codec Mandatory Setting (ex: G.711 only, etc) - Codec Priority Setting at Internal and External call	○
	Comfort Noise VAD (Voice Activity Detection) Echo Cancellation	○
	DTMF Processing -In-band and RFC2833 DTMF Support - RFC2976(SIP INFO Method) Out-of-band DTMF support	○
	FAX Relay using T.38 Real-time FAX Protocol(T.38)	○

IPNext IP-PBX Test Result

IP-PBX Supplementary Service

Classification	Remark	Result
IP-PBX Supplementary Service	Incoming Call Transfer - Unconditional Incoming call transfer - Incoming call transfer at busy - Incoming call transfer at no response	○
	Call Transfer at no response - No response time is changeable	○
	Call Status Change - Outbound Call Service Effect at Service Activation - Call Transfer Operation Check (Don't Care Terminal Mode or Status)	○
	Call waiting - New Arrival Call Receiving without disconnecting current call - Round robin call conversation	○
	Call Restriction depend on call type - Local Call, Long distance call, Internal Call, 060 etc	○
	Do Not Disturb Function	○
	Call Hold - Call Hold and Music on Hold at busy - Call Reconnection at Releasing Call Hold	○

IPNext IP-PBX Test Result

IP-PBX Supplementary Service

Classification	Remark	Result
IP-PBX Supplementary Service	Call Transfer - Blind Call Transfer - Consult Call Transfer	○
	Auto-Attendant - IVR announcement, IVR editor, etc - Extension line call receiving using DTMF - Incoming call receiving using Voice Mail - Call transfer to internal line or Group line	○
	Call Pickup - Pickup Group - Direct Call Pickup	○
	Service Activation/Deactivation using supplementary service function key	○
	Service Activation/Deactivation using a specific DTMF in case of no function key	○

IPNext IP-PBX Test Result

Voice Mail and Message

Classification	Remark	Result
VMS	Support Voice Mail Function	○
	Message Recording and Management	○
Announcement and Tone	Support announcement function to IP-PBX terminal (IP Phone, Video Phone, etc)	○
	Support Dial tone, Ring back tone, etc	○
	Busy tone or announcement support	○
Message	Message Send/Receive between IP-PBX terminals	○
	SMS message send/receive inter-working with KT Softswitch - SMS text (korean, english, special character) up to 80 byte - SMS broadcasting function - SMS message arrival alarm	○
System Capacity	IP-PBX terminal registration - More than 200 line	○
	E1 Interface - More than 2 Port	○
	Voice Mail Capacity - More than 6000minute	○

IPNext IP-PBX Test Result

Performance and Voice Quality

Classification	Remark	Result
Call Processing Performance	Call Processing Performance - Concurrent Call Processing : 50% of Proposed Capacity - Call Processing Power per second : 50 % of Concurrent call processing power 99 % Call Complete Rate during 24 hours : Measurement : Tektronics Sepctra2	○
	Simultaneous Call Processing when it use the external analog and Digital E1 gateway	○
Message Processing Performance	SMS message processing performance - message processing /sec : 2 Msg	○
	It should not be effect the call processing performance when IP-PBX process the SMS message	○
Voice Quality	G.711(A/u-Law) : more than MOS 4.0 G.729a : more than MOS 3.6 이상 Voice Delay: one-way 150ms under	○
System Redundancy (Duplication) & Stability	In case of main module fault, automatic fail over processing and maintain the IP-PBX service without system halt or service hold	○
	In case of main module's IP interface fault, automatic fail over processing and maintain the IP-PBX service without service hold	○

IPNext IP-PBX Test Result

System Management

Classification	Remark	Result
System Management	Support the GUI based Management Tool	○
	Configuration Management - Represent Number for authentication, domain, password - KT softswitch address and port number - User Account Registration, Rename, Delete - Group Account addition, rename, delete - System configuration such as dialplan	○
	Can generate Alarm when system fault is occurred Fault Management - System hardware fault - Software fault such as process, thread fault - Physical link error such as ethernet port, etc	○
	Statistic Management - Performance Statistic - Outbound and Inbound Call Statistic - Internal Call Statistic	○
	Security Management - User Management , Account Management (account delete, registration) - User Login/Logout	○
	Log Management - Call Log Support	○

IPNext IP-PBX Test Result

Upgrade and System Hardware

Classification	Remark	Result
System Upgrade	Upgrade Possibility for New Function and Bug Fix	○
	Maintain current service and feature after system upgrade	○
	When some problem is occurred during new version S/W upgrade, IP-PBX can return to current version S/W image and service	○
KT Neoss Interworking Function	Support SNMPv2 Management	○
	KT Neoss Inter-working	○
System Dimension and Power	19" Standard Rack Mount	○
	AC 220V or DC -48V(option)	○
	KT Requirement - Operating Temperature - Relative Humidity	○



Thank you!

AddPac Technology Co., Ltd.
Sales and Marketing

Phone +82.2.568.3848 (KOREA)

FAX +82.2.568.3847 (KOREA)

E-mail sales@addpac.com