AddPac
IP-PBX Solution

IP-PBX Solution

IPNext IP-PBX Solution

AddPac IPNext IP-PBX System is a next-generation IP-PBX System and a new IP telephony solution for all IP environment, mobile network (GSM, CDMA, 3G), and traditional PSTN network. AddPac IPNext IP-PBX inter-works with various IP terminals of AddPac (e.g. AP-VP300, AP-VP280, AP-VP250, AP-VP150, AP-VP120 IP Video Phones, and AP-IP300, AP-IP230, AP-IP160, AP-IP120, AP-IP90 IP Phones) to provide multimedia IP telephony services as well as the traditional legacy PBX features. These IP-PBX products are based on the advanced embedded RISC that enables firmware upgrade. AddPac IPNext IP-PBX solution is suitable for medium and large size companies, and inter-works with the VoIP and video products of AddPac Technology such as GSM Gateway, Video Conference System to provide a variety of IP application services appropriate for your network. AddPac IPNext IP-PBX System supports H.323 VoIP inter-working as well as SIP protocol for outbound call.

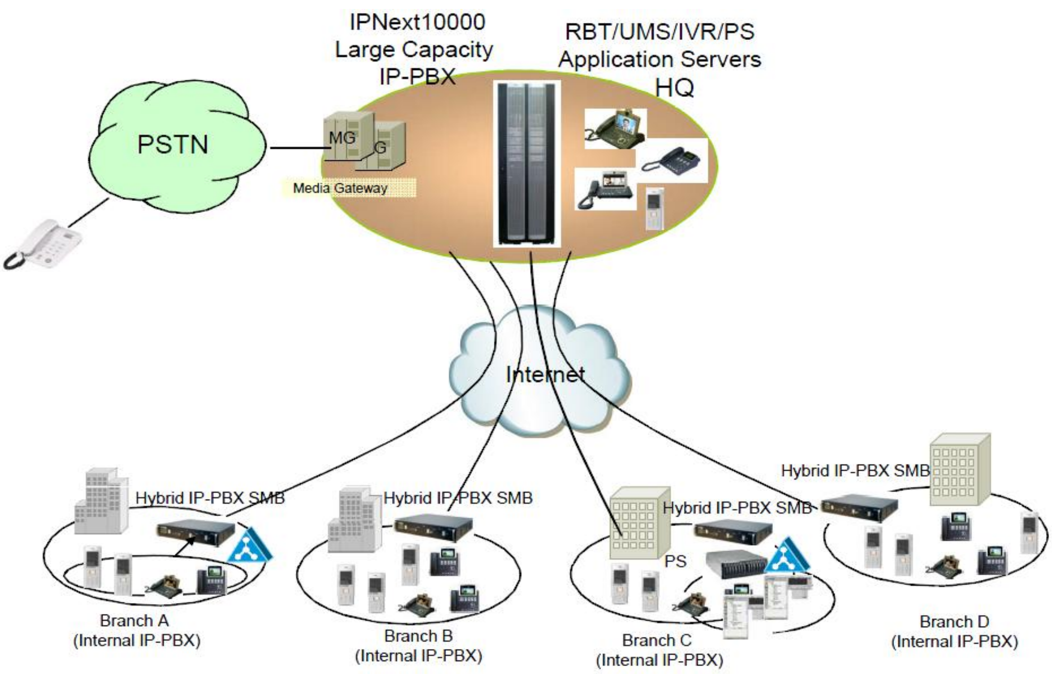
IPNextIP-PBX Comparison Table

Model / **Service Features**		IPNext10000 	IPNext5000 	IPNext3000 	IPNext2000 
Registration User Number		More than 10000	5000	3000	2000
Concurrent Call User Number		5000	1000	800	500
IPv4/IPv6 Dual Stack Support		Support	Support	Support	Support
VoIP Signaling	Internal	SIP	SIP	SIP	SIP
	External	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP
Powerful IVR, UMS, Media Service, User Presence Service		Support	Support	Support	Support
RTP Proxy Service (IPv6, Private IP)		Support	Support	Support	Support
LAN Port		2	2	2	2
System Duplication		Support	Support(built-in)	Support(built-in)	Support(built-in)

IPNext IP-PBX Comparison Table

Model Service Features		IPNext1000 	IPNext700 	IPNext600 	IPNext500 
Registration User Number		1000	700	500	500
Concurrent Call User Number		500	300	100	100
IPv4/IPv6 Dual Stack Support		Support	Support	Support	Support
VoIP Signaling	Internal	SIP	SIP	SIP	SIP
	External	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP
Powerful IVR, UMS, Media Service, User Presence Service		Support	Support	Support	Support
RTP Proxy Service (IPv6, Private IP)		Support	Support	Support	Support
LAN Port		2	2	2	2
System Duplication		Support	Support	Support(Built-In)	Support

Network Diagram



Large-Scale IP-PBX Solution

AddPac IPNext10000 Large-Scale IP-PBX System is a next-generation IP-PBX System and a new IP telephony solution for all IP environment, mobile network (GSM, CDMA, 3G), and traditional PSTN network. IPNext10000 IP-PBX inter-works with various IP terminals of AddPac (e.g. AP-VP300, AP-VP280, AP-VP250, AP-VP150, AP-VP120 IP Video Phones, and AP-IP300, AP-IP230, AP-IP160, AP-IP120, AP-IP90 IP Phones) and other vendor IP phones to provide multimedia IP telephony services as well as the traditional legacy PBX features. These Large-Scale IP-PBX products are based on the advanced high performance Intel Processor that enables firmware upgrade. AddPac Large-Scale IP-PBX solution is suitable for large size companies and telecom operators, and inter-works with the VoIP and video products of AddPac Technology such as GSM gateway, video conference system to provide a variety of IP application services appropriate for your network. IPNext10000 IP-PBX System supports H.323 VoIP inter-working as well as SIP protocol for outbound call, call manager service and IP centric service. AP-AS10000 Application Server supports various supplementary service as a companion system of IPNext10000 IP-PBX like as User Presence Service, IVR Service, RBT(Ring Back Tone) Service, UMS(Unified Messaging) Service, etc.

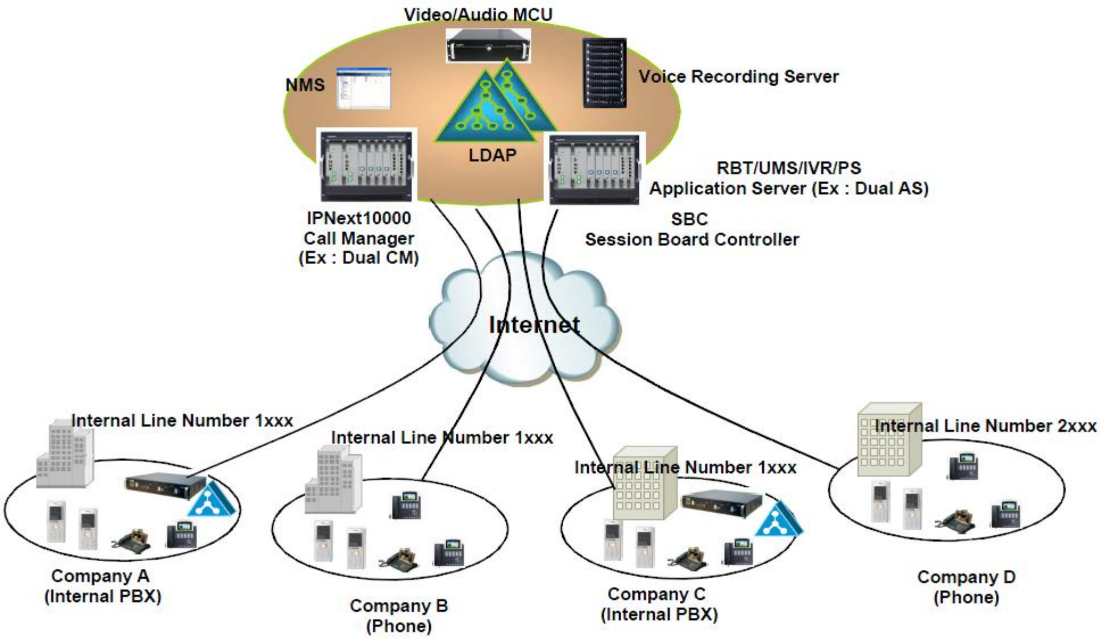
IPNext10000N Large-Scale IP-PBX

Model **Service Features**		IPNext10000N 
Registration User Number		More than 10000
Concurrent Call User Number		5000
IPv4/IPv6 Dual Stack Support		Support
VoIP Signaling	Internal	SIP
	External	H.323/SIP
Powerful IVR, UMS, Media Service, User Presence Service		Support
RTP Proxy Service (IPv6, Private IP)		Support
LAN Port		1
System Duplication		Support

AS10000N Application Server for Large-Scale IP-PBX

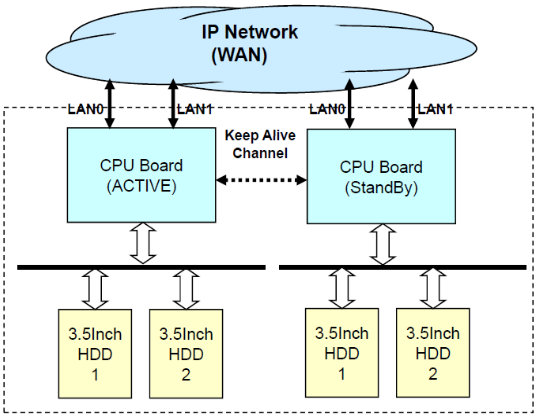
Model Service Features	AP-AS10000N 
Presence Server	Provides User Extension Presence Monitoring Service
	Support Directory Service to Smart Messenger and Terminals
IVR (Interactive Voice Response) Service	Provides Various IVR Scenarios Made by IVR Editor
	Support Schedule based Scenario
RBT (Ring Back Tone) Service	Provides Department and Personal based MoH and RBT
	Provides Announcement and Tone Package Management
UMS (Unified Messaging) Service	Provides Voice Mail Store and Play
	Provides SMS Delivery Service

Network Diagram



Dual Redundancy IP-PBX Solution

AddPac Technology's Dual Redundancy IP-PBX Solution is a new fault tolerant IP telephony solution for all IP environment, mobile network (GSM, CDMA, 3G), and traditional PSTN network. Dual Redundancy IP-PBX system provide one-system dual IP-PBX system shown as following diagram



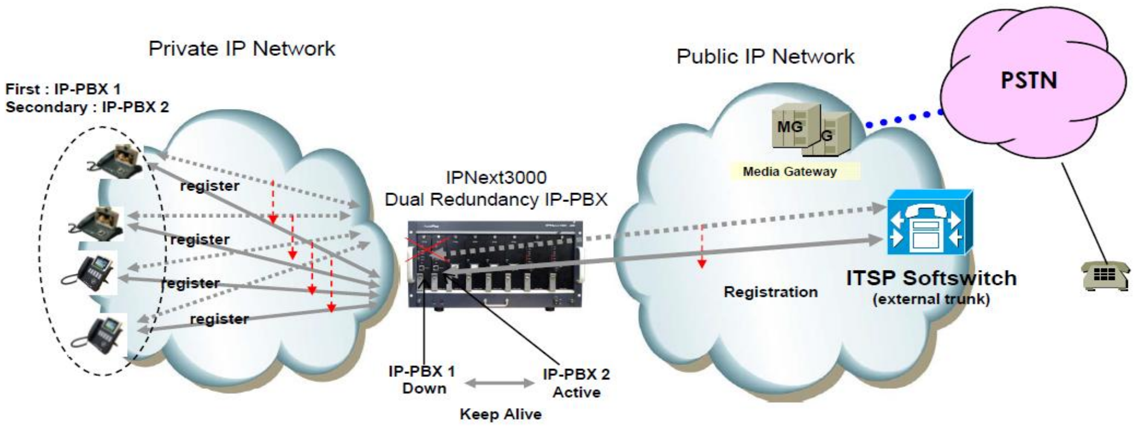
One System Dual Redundancy IP-PBX system Block Diagram

Dual Redundancy IP-PBX system supports the dual CPU boards and adopts the dual HDD system per CPU boards for hard disk redundancy (RAID1 configuration). Also, Dual Redundancy IP-PBX supports the dual internal module type power supply. It's ideal IP-PBX system for medium and large businesses taking a full advantage of system stability. Dual Redundancy IP-PBX system inter-works with various IP terminals of AddPac (e.g. AP-VP300, AP-VP280, AP-VP250, AP-VP150, AP-VP120 IP Video Phones, and AP-IP300, AP-IP230, AP-IP160, AP-IP120, AP-IP90 IP Phones) to provide multimedia IP telephony services as well as the traditional legacy PBX features. These IP-PBX products are based on the advanced embedded RISC that enables firmware upgrade. AddPac Dual Redundancy IP-PBX systems are suitable for medium and large size companies, and inter-works with the VoIP and video products of AddPac Technology such as GSM gateway, video conference system to provide a variety of IP application services appropriate for your network. AddPac Dual Redundancy IP-PBX system supports H.323 VoIP inter-working as well as SIP protocol for outbound call.

Dual Redundancy IP-PBX Comparison Table

Model / Service Features		IPNext5000 	IPNext3000 	IPNext2000 	IPNext600 
Registration User Number		5000	3000	2000	500
Concurrent Call User Number		1000	800	500	100
IPv4/IPv6 Dual Stack Support		Support	Support	Support	Support
VoIP Signaling	Internal	SIP	SIP	SIP	SIP
	External	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP
Powerful IVR, UMS, Media Service, User Presence Service		Support	Support	Support	Support
RTP Proxy Service (IPv6, Private IP)		Support	Support	Support	Support
LAN Port		2	2	2	2
System Duplication		Support(built-in)	Support(built-in)	Support(built-in)	Support(Built-In)

Network Diagram



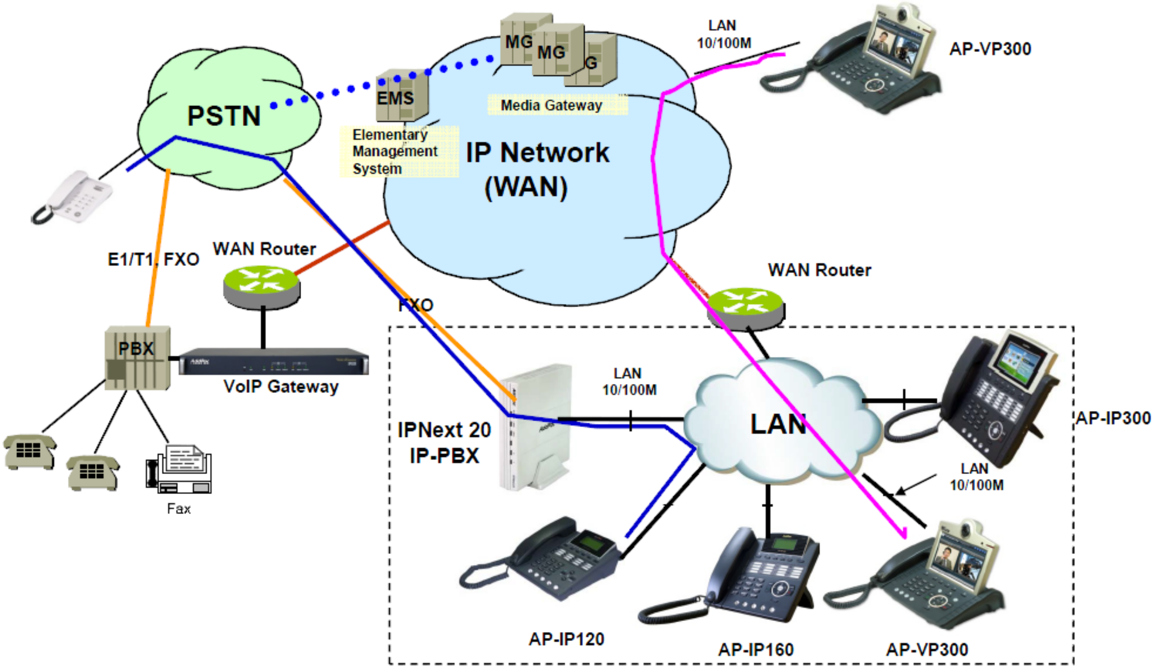
SOHO IP-PBX Solution

AddPac SOHO IP-PBX is a next-generation IP-PBX and a new IP telephony solution for all IP environments and traditional PSTN network. SOHO IP-PBX inter-works with various IP terminals of AddPac (e.g. AP-VP500, AP-VP300, AP-VP280, AP-VP250, AP-VP230, AP-VP150, AP-VP120 IP Video Phone, and AP-IP300, AP-IP160, AP-IP120, AP-IP90 IP Phone) to provide multimedia IP telephony services as well as the traditional Legacy PBX features. This SOHO IP-PBAX product is based on the advanced embedded RISC that enables firmware upgrade, and can be equipped with PSTN VoIP interfaces (FXO) depending on options. AddPac SOHO IP-PBX system is suitable for small companies, and inter-works with the VoIP and video products of AddPac Technology to provide a variety of IP application services appropriate for your network. Also, SOHO IP-PBX supports H.323 VoIP inter-working as well as SIP protocol for outbound call.

AddPac SOHO IP-PBX Comparison Table

Model / Service Features		IPNext150 	IPNext100 	IPNext50 	IPNext20 
Registration User Number		70	50	30	10
Concurrent Call User Number		35	20	15	5
IPv4/IPv6 Dual Stack Support		Support	Support	Support	Support
VoIP Signaling	Internal	SIP	SIP	SIP	SIP
	External	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP
Powerful IVR, UMS, Media Service, User Presence Service		Support	Support	Support	Support
RTP Proxy Service (IPv6, Private IP)		Support	Support	Support	Support
LAN Port		2	2	2	2
USB Interface		N/A	Support	Support	Option
FXO, FXS Interface		N/A	Option	Option	Option

Network Diagram



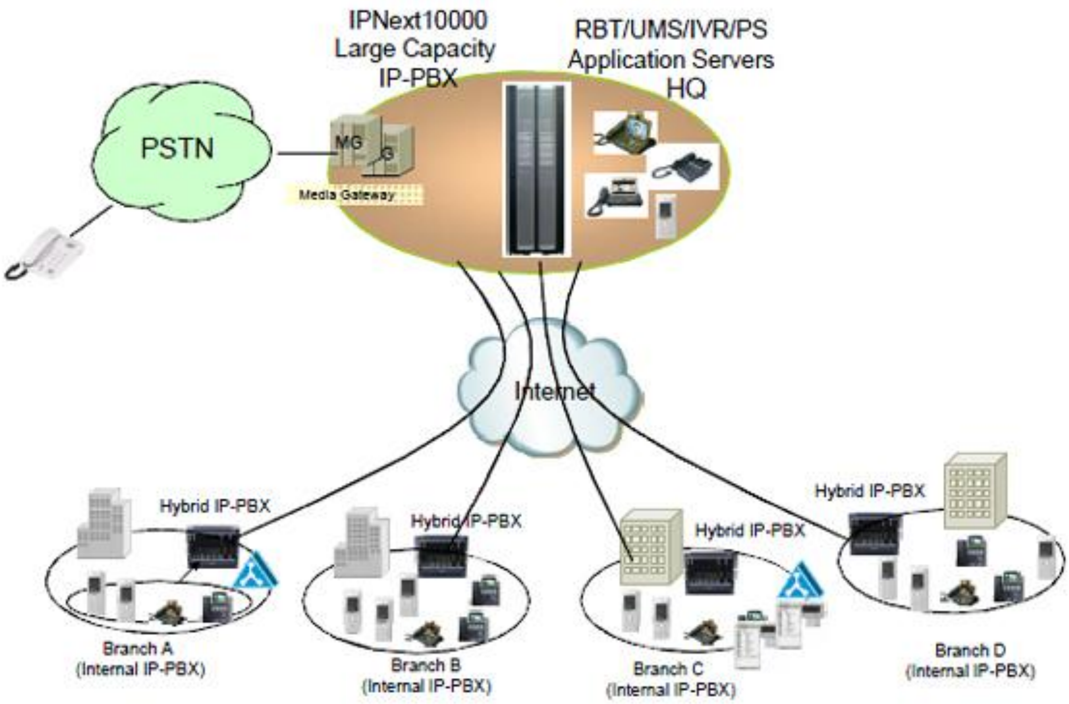
Large Capacity Hybrid IP-PBX Solution

AddPac Large Capacity Hybrid IP-PBX Systems are next-generation IP-PBX Systems for all IP environment and traditional PSTN network. Hybrid IP-PBX inter-works with various IP terminals of AddPac (e.g. AP-VP300, AP-VP280, AP-VP250, AP-VP150, AP-VP120 IP Video Phones, and AP-IP300, AP-IP230, AP-IP160, AP-IP120, AP-IP90 IP Phones) and analog phones(low cost compared with IP terminals, also reused) to provide multimedia IP telephony services as well as the traditional legacy PBX features. This Hybrid IP-PBX product is based on the advanced embedded RISC that enables firmware upgrade, and can be equipped with various VoIP interfaces (FXS, FXO, Digital E1/T1, etc) depending on module options. AddPac Large Capacity Hybrid IP-PBX solution is suitable for medium size enterprise companies, and inter-works with the VoIP and video products of AddPac Technology to provide a variety of IP application services appropriate for your network.

Large Capacity Hybrid IP-PBX Comparison Table

Model / Service Features		IPNext350 	IPNext320 
Registration User Number		500	350
Concurrent Call User Number		150	100
IPv4/IPv6 Dual Stack Support		Support	Support
VoIP Signaling	Internal	SIP	SIP
	External	H.323/SIP	H.323/SIP
Powerful IVR, UMS, Media Service, User Presence Service		Support	Support
RTP Proxy Service(IPv6, Private IP)		Support	Support
LAN Port (Gigabit)		2	2
VoIP Module Slots for PSTN		8 Slots x (32-Port Module) = 256 Ports	4 Slots x (32-Port Module) = 128 Port
VoIP Interface		FXS, FXO, etc	FXS, FXO, etc

Network Diagram



Hybrid IP-PBX Solution

AddPac Hybrid IP-PBX Systems is a next-generation IP-PBX System and a new IP telephony solution for all IP environment and traditional PSTN network. Hybrid IP-PBX inter-works with various IP terminals of AddPac (e.g. AP-VP300, AP-VP280, AP-VP250, AP-VP150, AP-VP120 IP Video Phones, and AP-IP300, AP-IP230, AP-IP160, AP-IP120, AP-IP90 IP Phones) and analog phones(low cost compared with IP terminals, also reused) to provide multimedia IP telephony services as well as the traditional legacy PBX features. This Hybrid IP-PBX product is based on the advanced embedded RISC that enables firmware upgrade, and can be equipped with various VoIP interfaces (FXS, FXO, Digital E1/T1, etc) depending on module options. AddPac Hybrid IP-PBX solution is suitable for small and medium size companies, and inter-works with the VoIP and video products of AddPac Technology to provide a variety of IP application services appropriate for your network. Also, AddPac Hybrid IP-PBX System supports H.323 VoIP inter-working as well as SIP protocol for outbound call.

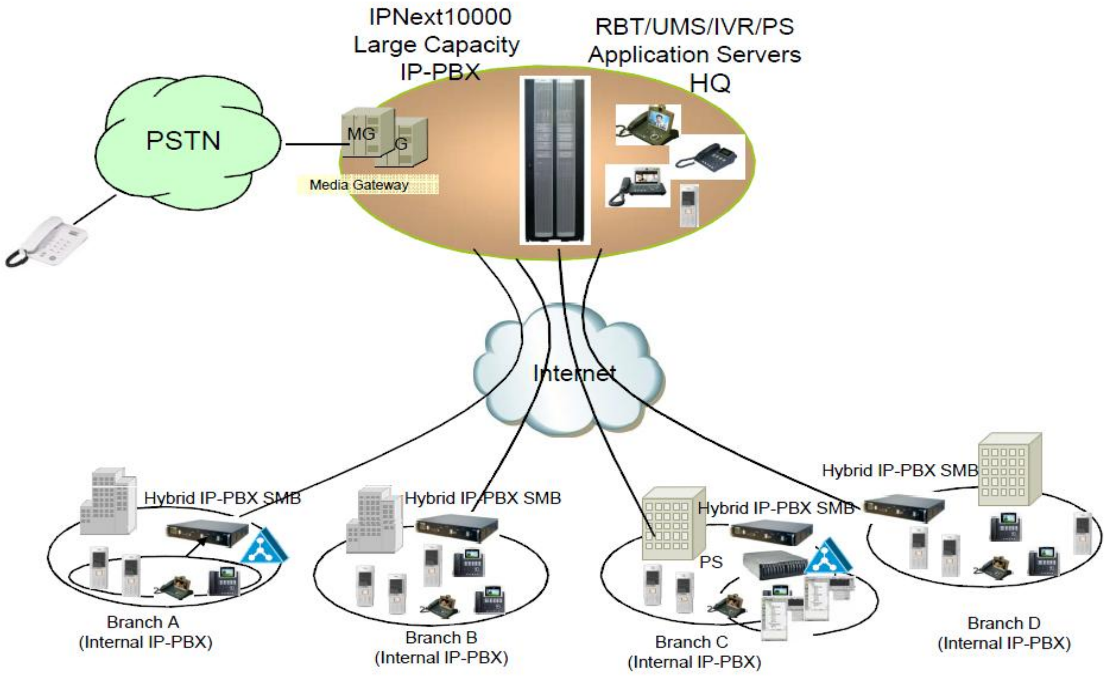
AddPac Hybrid IP-PBX Comparison Table

Model / Service Features		IPNext350 	IPNext320 
Registration User Number		500	350
Concurrent Call User Number		150	100
IPv4/IPv6 Dual Stack Support		Support	Support
VoIP Signaling	Internal	SIP	SIP
	External	H.323/SIP	H.323/SIP
Powerful IVR, UMS, Media Service, User Presence Service		Support	Support
RTP Proxy Service(IPv6, Private IP)		Support	Support
LAN Port (Gigabit)		2	2
VoIP Module Slots for PSTN		8 Slots x (32-Port Module) = 256 Ports	4 Slots x (32-Port Module) = 128 Port
VoIP Interface		FXS, FXO, etc	FXS, FXO, etc

Model Service Features		IPNext280PTT 	IPNext250 	IPNext230 	IPNext210 
Registration User Number		300	200	200	150
Concurrent Call User Number		100	100	60	50
IPv4/IPv6 Dual Stack Support		Support	Support	Support	Support
VoIP Signaling	Internal	SIP	SIP	SIP	SIP
	External	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP
Powerful IVR, UMS, Media Service, User Presence Service		Support	Support	Support	Support
RTP Proxy Service (IPv6, Private IP)		Support	Support	Support	Support
LAN Port		2	2	2	2
VoIP Module Slots for PSTN		4 Slots (8 Port Module)	10 Slots (8 Port Module)	15 Slots (4 Port Module)	4 Slots (8 Port Module)
VoIP Interface		FXS, FXO,E&M, E1/T1	FXS, FXO,E1/T1	FXS, FXO, E&M,E1/T1	FXS,FXO,E&M,E1/T1

Model Service Features		IPNext190 	IPNext187 	IPNext185 	IPNext180 
Registration User Number		150	150	240	100
Concurrent Call User Number		50	50	120 (Up to 4E1)	30
IPv4/IPv6 Dual Stack Support		Support	Support	Support	Support
VoIP Signaling	Internal	SIP	SIP	SIP	SIP
	External	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP
Powerful IVR, UMS, Media Service, User Presence Service		Support	Support	Support	Support
RTP Proxy Service (IPv6, Private IP)		Support	Support	Support	Support
LAN Port		2	2	2	2
VoIP Module Slots for PSTN		4 Slots (8 Port Module)	3 Slots (8 Port Module)	2 Slots (8 Port Module)	2 Slots (8 Port Module)
VoIP Interface		FXS,FXO,E1/T1	FXS,FXO,E1/T1	FXS,FXO,E1/T1	FXS,FXO,E1/T1

Network Diagram



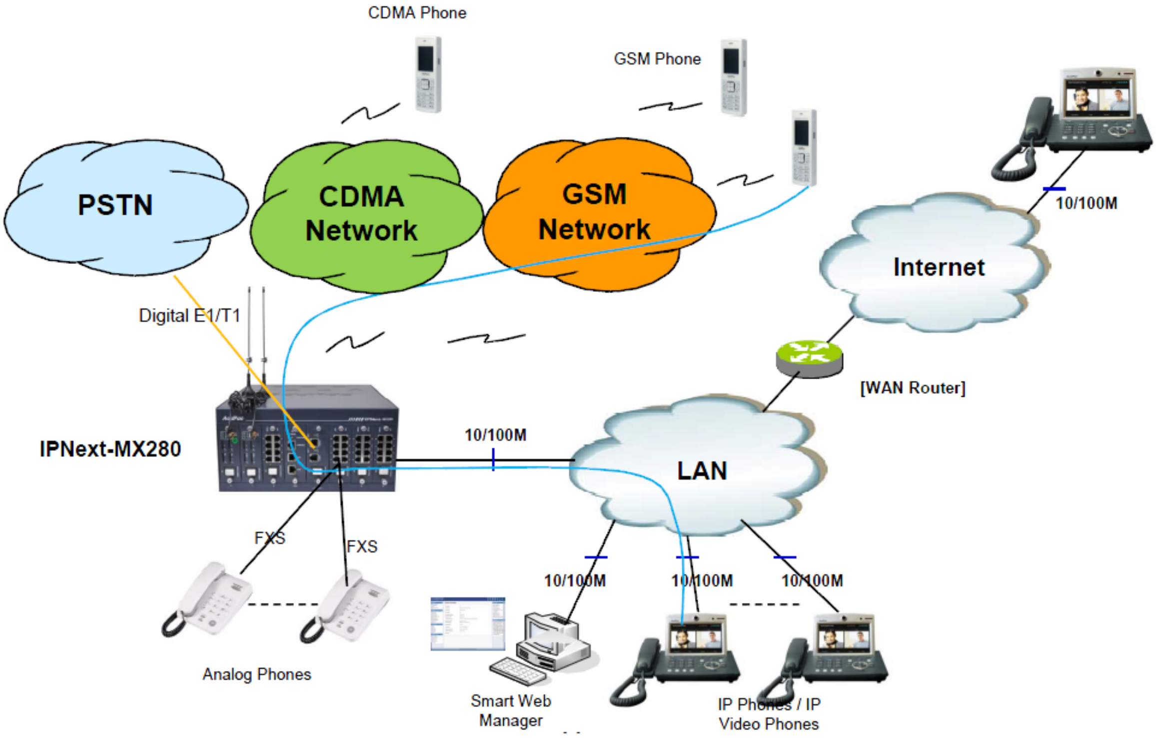
Mobile Hybrid IP-PBX Solution

AddPac mobile hybrid IP-PBX Systems is a next-generation IP-PBX System and a new IP telephony solution for all IP environment, mobile network(GSM, CDMA, 3G, etc), and traditional PSTN network. Mobile hybrid IP-PBX inter-works with various IP terminals of AddPac (e.g. AP-VP300, AP-VP280, AP-VP250, AP-VP150, AP-VP120 IP Video Phones, and AP-IP300, AP-IP230, AP-IP160, AP-IP120, AP-IP90 IP Phones) and analog phones(low cost compared with IP terminals, also reused) to provide multimedia IP telephony services as well as the traditional legacy PBX features. This mobile hybrid IP-PBX product is based on the advanced embedded RISC that enables firmware upgrade, and can be equipped with various VoIP interfaces (GSM, CDMA, 3G, FXS, FXO, Digital E1/T1, etc) depending on module options. AddPac mobile hybrid IP-PBX solution is suitable for small and medium size companies, and inter-works with the VoIP and video products of AddPac Technology to provide a variety of IP application services appropriate for your network. Also, AddPac mobile hybrid IP-PBX System supports H.323 VoIP inter-working as well as SIP protocol for outbound call.

Mobile Hybrid IP-PBX Comparison Table

Model / Service Features		IPNext-MX280 	IPNext-MX260 	IPNext-MX250 
Registration User Number		300	200	150
Concurrent Call User Number		100	70	50
IPv4/IPv6 Dual Stack Support		Support	Support	Support
VoIP Signaling	Internal	SIP	SIP	SIP
	External	H.323/SIP	H.323/SIP	H.323/SIP
Powerful IVR, UMS, Media Service, User Presence Service		Support	Support	Support
RTP Proxy Service (IPv6, Private IP)		Support	Support	Support
LAN Port		2	2	2
VoIP Module Slots for Mobile, PSTN		7 Slots	5 Slots	4 Slots
Mobile Interface				
VoIP Interface		FXS, FXO,E&M, E1/T1	FXS, FXO,E1/T1	FXS,FXO,E&M,E1/T1

Network Diagram



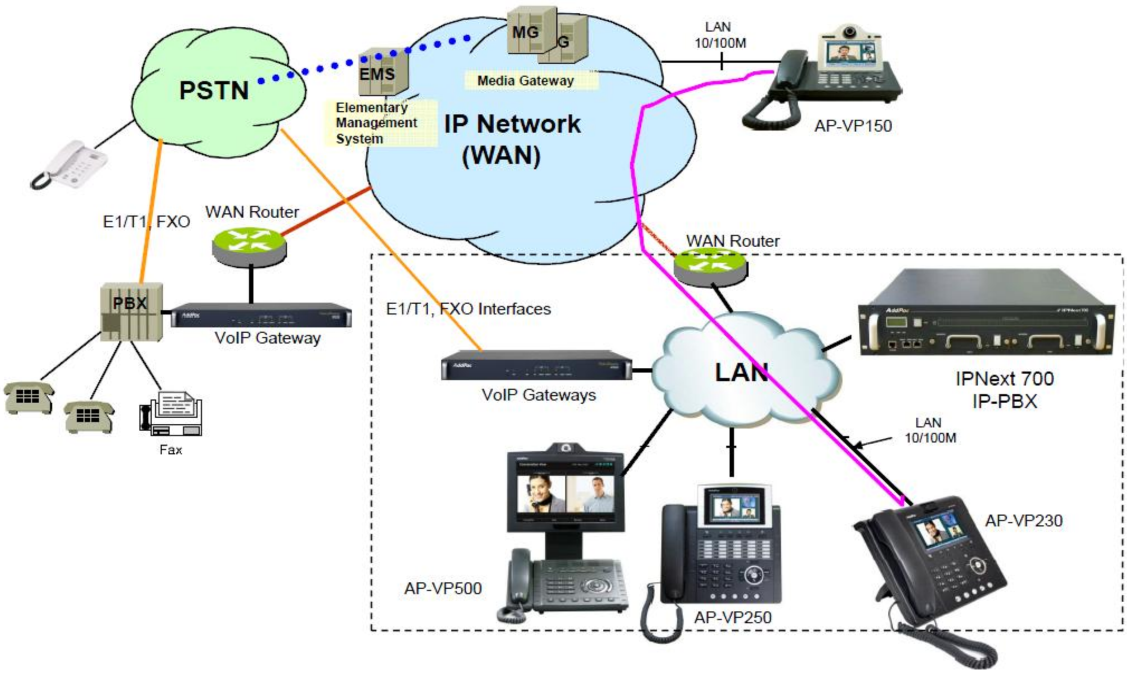
Video Phone Solution

AddPac Technology IP video phone series are designed to deliver high quality video telephony service over IP network. This new and potent video phones deliver the various IP communication solutions taking a full advantage of new 'all-in-one concept' with voice, audio and video application integrated. It provides not only cutting-edge features such as various AV in/out interfaces, QoS functions, public IP sharing but also a wide range of multiple VoIP signaling such as SIP, H.323 protocols and H.263, MPEG4, H.264 video codecs. With its modern design, AddPac Video Phones are equipped with the latest audio/video codec, high quality LCD and Built-In Camera interfaces. From the video conferencing, video telephony to the communication aid for the disabled, AddPac Video Phone becomes the best choice of all.

IP Video Phone Comparison Table

	AP-VP500 	AP-VP350 	AP-VP300N 	AP-VP280 	AP-VP250 	AP-VP230 	AP-VP150 	AP-VP120 
LCD Size	12.1 Inch Touch Screen	7Inch Touch Screen	7Inch Touch Screen	7Inch Touch Screen	4.3Inch Touch Screen	5Inch Touch Screen	4.3Inch Touch Screen	4.3Inch
Camera	CCD	CCD	CCD	CMOS	CMOS	CMOS	CCD	CMOS
Video Codec	H.263 MPEG4 H.264	H.263 MPEG4 H.264	H.263 MPEG4 H.264	H.263 MPEG4 H.264	H.263 MPEG4 H.264	H.263 MPEG4 H.264	H.263 MPEG4 H.264	H.263 MPEG4 H.264
Signaling	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP
Video MCU	N/A	4-Party Video MCU	N/A	N/A	N/A	N/A	N/A	N/A
Voice MCU	3-Party	3-Party	3-Party	3-Party	3-Party	3-Party	3-Party	3-Party
LAN Port	2	2	2	2	2	2	2	2
PoE	N/A	N/A	Support	N/A	Support	Support	Support	Support

Network Diagram



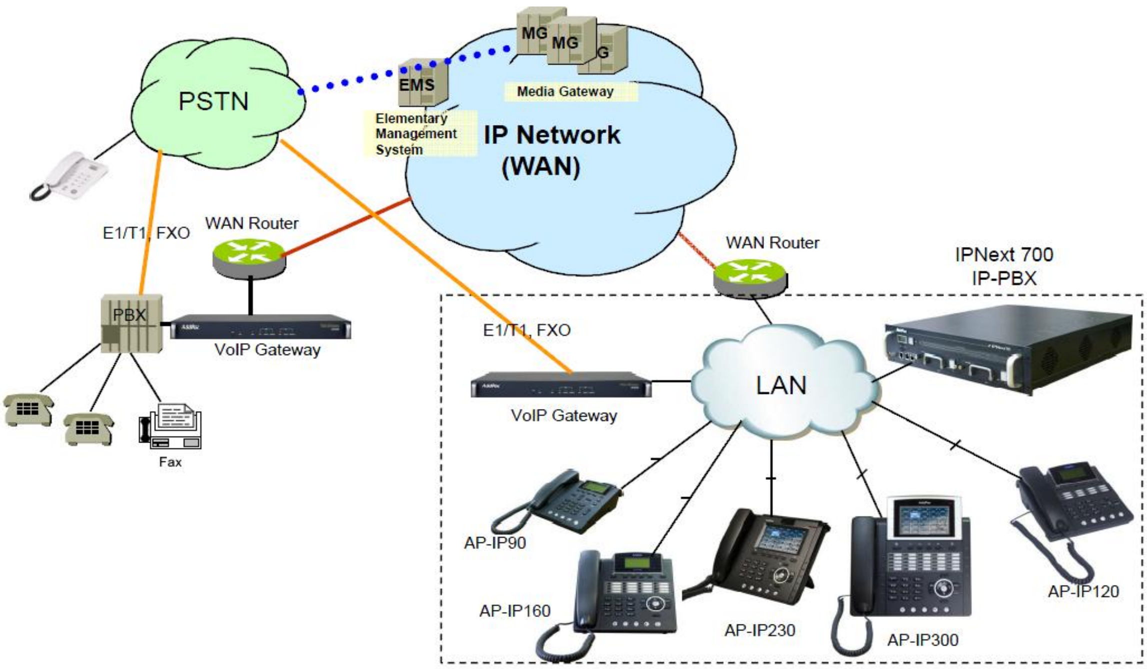
IP Phone Solution

AddPac Technology IP phones are designed to provide enhanced IP telephony functionality to meet the wide range of business user requirements. This IP telephones optimally deliver rich featured voice telephony service on ordinary internet infrastructure as well as AddPac IP-PBX environment on local LAN as a fully featured IP extension for the complete AddPac VoIP solution. The new and versatile AddPac IP phone bring the integrated solution for the IP based voice communication and the broadcasting feature to maximize business potentials. It provides feature keys, customizable hot-keys, two(2) ethernet ports, the latest QoS, public IP sharing. It supports not only the major VoIP signaling protocols such as SIP, H.323, MGCP but also G.711, G.726 voice codec, stereo audio in/out interfaces for external Headset MIC, etc AddPac IP phones combine AddPac's field proven VoIP technology and IP voice broadcasting technology. Apart from telephony service, it delivers IP voice broadcasting service supporting external MIC/Line-in, Line-out interface for various input/output devices such as headset, Amp or speaker. Since AddPac IP phones support various voice codecs according to bandwidth environment, it can be deployed anywhere on the internet, ensuring optimal voice quality by leveraging the latest QoS technology. Furthermore, installed along with AddPac's comprehensive Hybrid IP=PBX, Call Manager, it not only improves operation offering an wide variety of features such as Music on Hold, Coloring service, Call Transfer but also provides the easy-to-use, intelligent IP telephony service enhanced by AddPac's unique PC-based User Agent.

IP Phone Comparison Table

	AP-IP300 	AP-IP250 	AP-IP230 	AP-IP160 	AP-IP120 	AP-IP90 
LCD Size	4.3 Inch Color LCD	4.3 Inch Color LCD	5 Inch Color LCD	4 Text Line Graphic LCD	4 Text Line Graphic LCD	4 Text Line Graphic LCD
Touch Screen	N/A	Support	Support	N/A	N/A	N/A
Speed-Dial Keys	25 Key with Presence LED	Touch Screen based 25 Keys	Touch Screen based 25 Keys	16 Key with Presence LED	12 Key with Presence LED	N/A
Voice Codec	G.711/G.726/ G.729/G.723	G.711/G.726/ G.729/G.723	G.711/G.726/ G.729/G.723	G.711/G.726/ G.729/G.723	G.711/G.726/ G.729/G.723	G.711/G.726/ G.729/G.723
Signaling	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP
3-Party Conversation	Support	Support	Support	Support	Support	Support
LAN Port	2	2	2	2	2	2
PoE(Option)	Support	Support	Support	Support	Support	Support
FXO(Option)	Support	Support	Support	Support	Support	Support

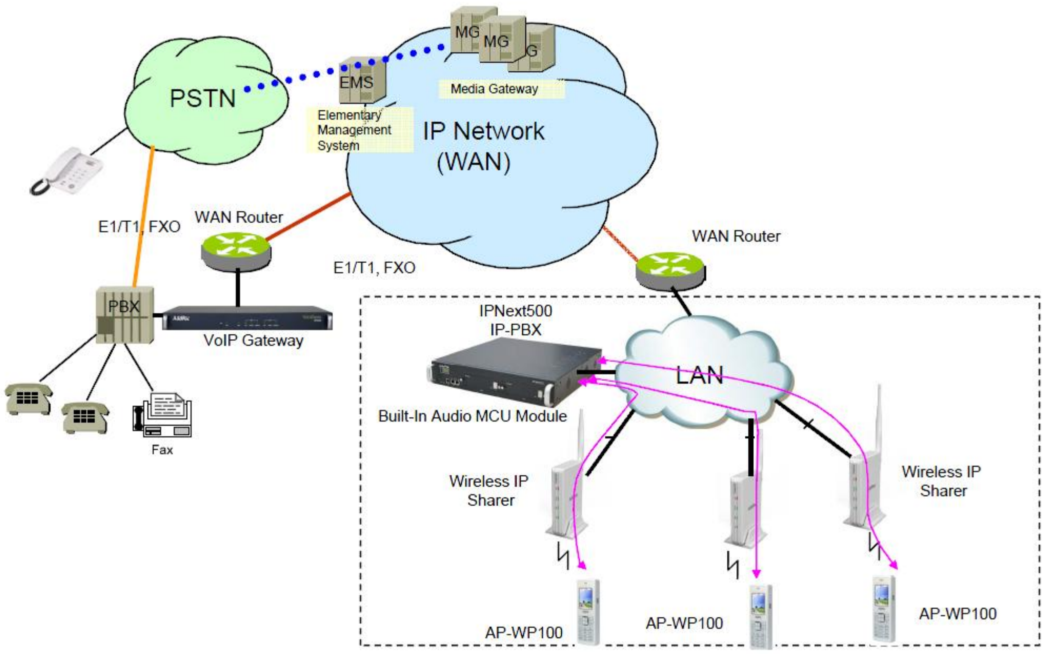
Network Diagram



WiFi IP Phone Solution

AP-WP100 WiFi IP Phone is the high performance and multi-functional wireless IP Phone with the advanced voice processing technology, which AddPac has been accumulated from many different areas of VoIP and IP Broadcasting and added it to IP Voice Broadcasting Technology. AP-WP100 is integrated with AddPac Technology IP telephony and broadcasting service solution. This Wi-Fi IP Phone provides the advanced features and service of SIP-based VoIP signaling protocol, the voice codec of G.711, stereo audio with input/output interface for headset, large-sized graphic LCD, a variety of function keys (Push-to-Talk Button), 802.11 b/g wireless interface, power adapter and USB interface, QoS, broadcasting and public IP-sharing. To cope with many different delays in Internet and packet error in the condition of low-bandwidth, QoS and error concealment is used to guarantee the optimal voice quality. Internetworking with a variety of the AddPac IP telephony solution (IP-PBX, MCU, Video Codec, VoIP Gateway, Mobile Gateway, IP Terminals and others) and the PC-based smart messenger, you can enjoy the convenient and intelligent IP telephony service.

Network Diagram



IP Soft Video Phone Solution

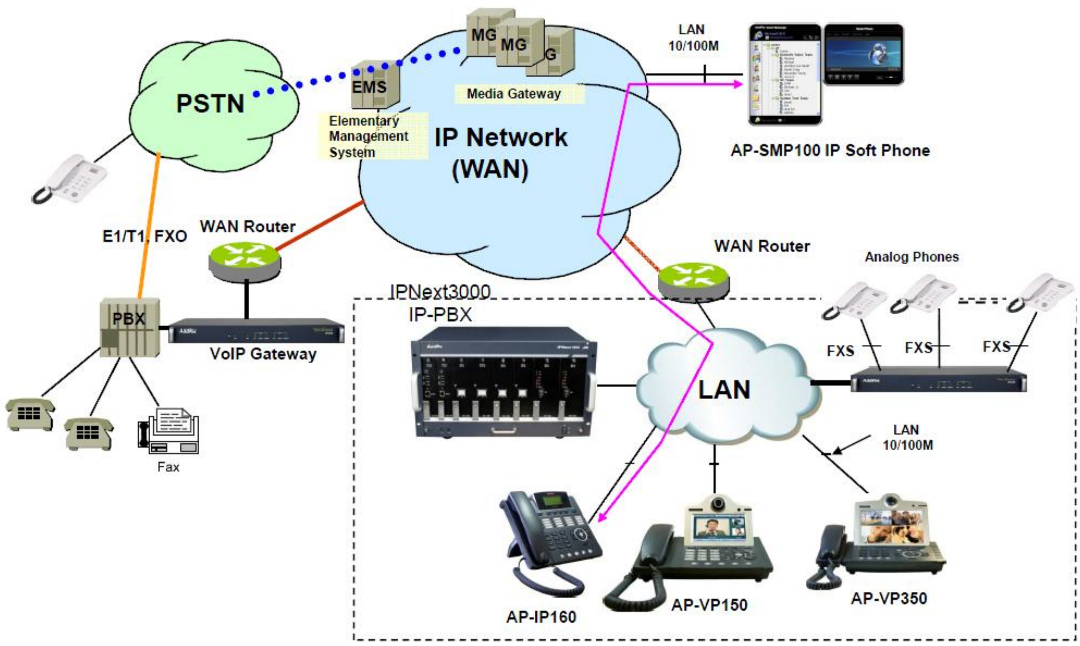
AP-SMP100 IP soft video phone is a MS Window based smart multimedia soft phone that allows you to make high quality and high resolution video communications by using the Internet. This product supports a video conference, point-to-point video communication, and aid to audio/video communication access for the disabled optimally through the latest audio/video codecs and intelligent smart messenger capability. The criteria of choosing a soft phone are high quality audio/video service and easy dial-access features. AP-SMP100 is a new IP based multimedia soft phone where Internet voice communications are combined with Internet video communications. This product provides advanced features and services such as video codecs (e.g. H.263, MPEG-4, and H.264), QoS, SIP VoIP signaling protocol, intelligent smart messenger, SSCP (smart service control protocol: AddPac Proprietary Protocol) for IP-PBX interworking, Video MCU interworking service for video conference. AP-SMP100 is a high-performance multi-functional IP soft phone where the state-of-the-art video processing technology is added to the voice processing technologies developed by AddPac in the VoIP field. Since AP-SMP100 supports networks of 64 Kbps to several Mbps, it is available at any Internet-enabled place. Above all, AP-SMP100 ensures the best video quality due to the ‘rate control’ function that ensures the best video quality and frame rate at a limited bandwidth and the high-end error resilience technology for troubleshooting various packet failures on the Internet.

IP Soft Video Phone Main View

AP-SMP100 Main Features

 AP-SMP100	Main Features	MS-Window based IP Soft Video Phone
		IP Real-time Audio/Video Broadcasting Terminal Solution
		Up to 30fps with VGA-Resolution(MPEG-4)
		Video Conference Call Support (AddPac External MCU Inter-working) Advanced Voice/Video Traffic QoS
		Advanced Voice/Video Traffic QoS
		SIP, H.323* Signaling Support
		Support Various Call Signal via AddPac IP-PBX Inter-working
	A/V Service Features	High-performance Video/Voice Codec Support - H.263, MPEG-4, JPEG, and H.264 - G.711,G.726
		Powerful Image Resolution Support -QCIF(176x144), CIF(352x288), QVGA(320x240), 4CIF and VGA
		64Kbps to 4Mbps Operating Video Traffic Bandwidth
		Rate Control for Video Traffic QoS Ensuring Optimized Quality Frame Rate with Limited Bandwidth
		High-end Error Resilient Against Various Packet Error

Network Diagram



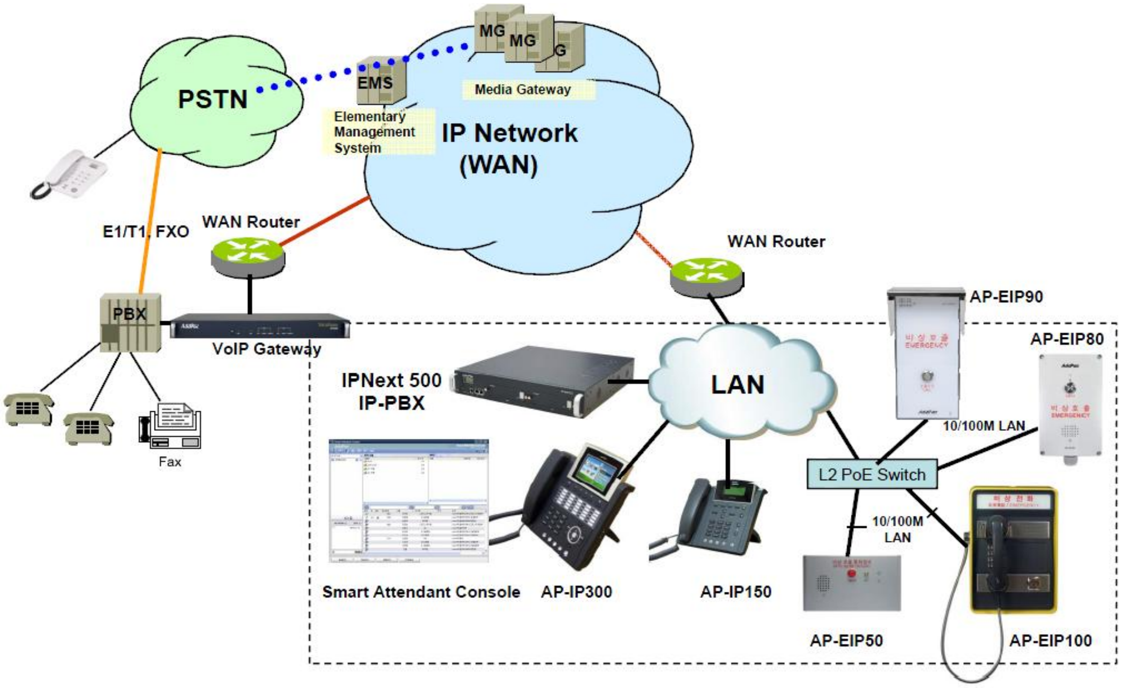
IP Emergency Call Phone Solution

AddPac IP based emergency phones(AP-EIP100, AP-EIP80, AP-EIP50,etc) enable the high-quality voice communications by using internet. This phone can be used in many different places requiring emergency call features and supports voice telephony service at optimum level in the general internet environment and the environment of the AddPac IP-PBX solution such as ADSL through all different kinds of transceiver, 3 x 4 dial button, voice codec, speaker interface and Power over Ethernet (PoE). Many features of AddPac IP emergency phone are highly distinguished and they have been designed in high voice quality DSP-based hardware architecture and integrated with Acoustic Echo Canceller which supports the built-in microphone and 2-way communication of Full Duplex. Also IP emergency phone supports SIP VoIP signaling protocols and a wide range of voice codec of G711, G726, G729 and G.723.1 depending on bandwidth availability, so AddPac IP emergency phones can be used in just about anywhere with internet connection. This emergency telephone has been implemented with QoS to comply with the packet error and all different kinds of latency on internet, especially in a limited bandwidth situation

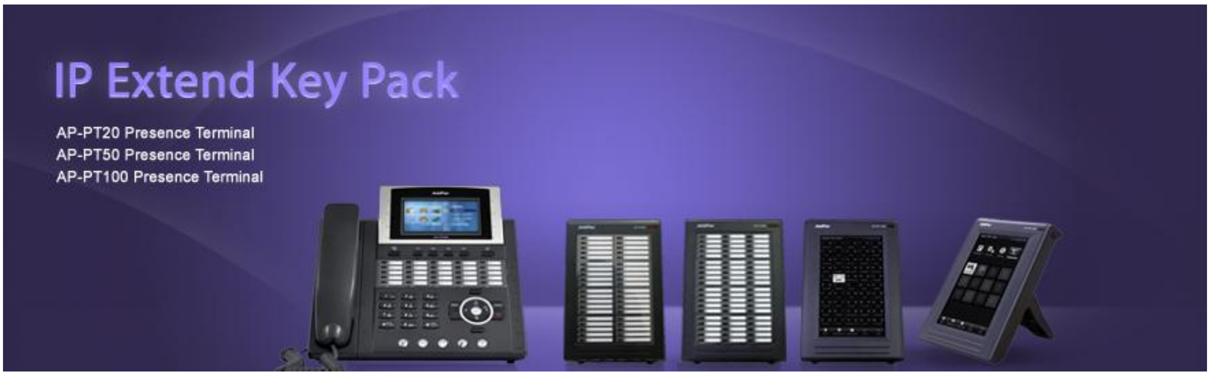
AddPac Emergency Call IP Phone Terminal Comparison Table

Model Service Features	AP-EIP100 	AP-EIP90 	AP-EIP80 	AP-EIP50 
Duplex	Full Duplex	Full Duplex	Full Duplex	Full Duplex
Key Pad	3x4 Key Support	N/A	N/A	N/A
Handset	Support	N/A	N/A	N/A
Voice Codec	G.711/G.726/ G.729/G.723	G.711/G.726/ G.729/G.723	G.711/G.726/ G.729/G.723	G.711/G.726/ G.729/G.723
Signaling	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP
Speaker Phone	Support	Support	Support	Support
LAN Port	1	1	1	1
PoE(Option)	Support	Support	Support	Support
Application	Indoor	Outdoor(water proof)	Outdoor(water proof)	Indoor

Network Diagram

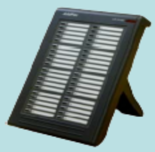


IP Extend Key Pack Terminal Solution

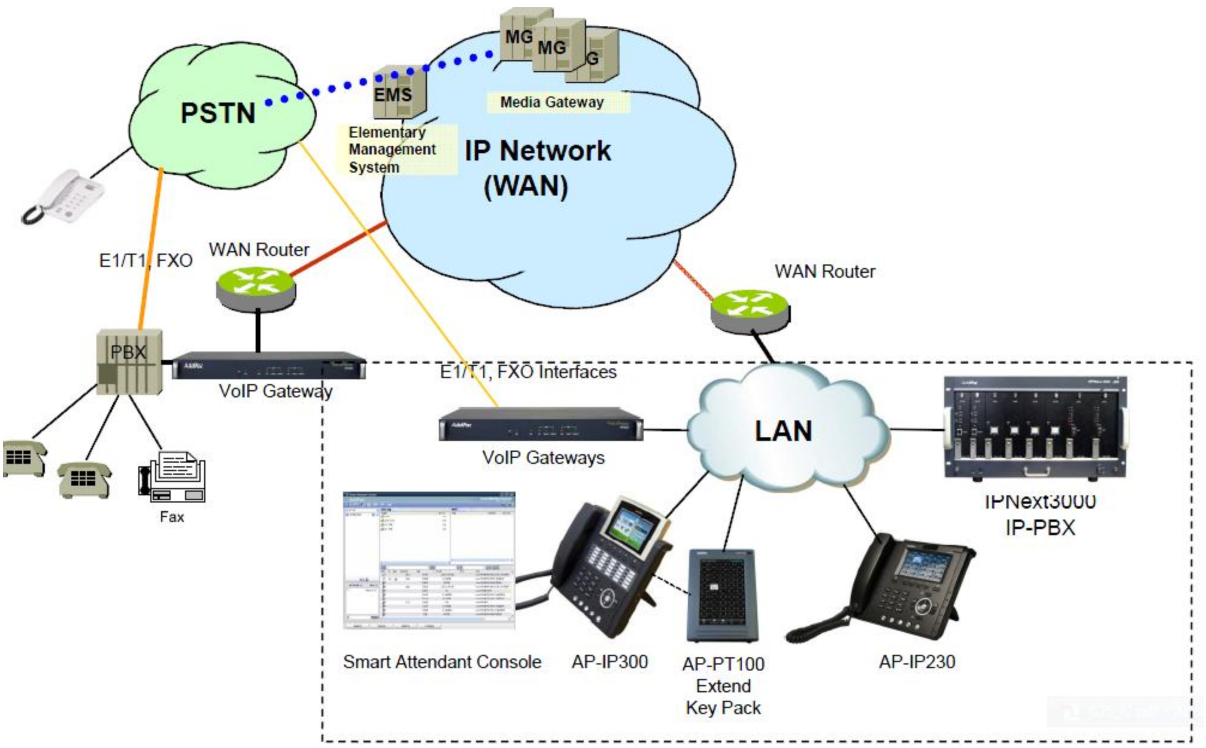


AddPac Technology's IP based Presence Terminal(Extend Speed-Dial Key Pack: AP-PT00, AP-PT50, AP-PT20) that interoperates with AddPac IP-PBX products, which are based on its next generation multi-media IP telephony solution. This product together with AddPac Video phone and IP phone can be used for User Presence Service and Speed Dialing Extend Key Pack Terminal solution. AddPac IP Extend Key Pack Terminals can be operated by AddPac Technology's proprietary protocol called Smart Service Control Protocol (SSCP), which is protocol for management between AddPac's IP PBX and IP CPE's. In order to provide User Presence features on real-time basis, such as user busy, on-line, dialing, etc, the User Presence server can be mounted and operated on AddPac IP-PBX. The User Presence server can be operated on the same hardware platform for SMB(small and medium business) solution and can be operated independently in the platforms separated to each other for large enterprise. Design on Ethernet based communication for interconnection between IP Phone/Video Phone terminal and IP Extend Key Pack Terminals or between IP Extend Key Pack Terminals and Presence Server, IP Extend Key Pack Terminals support flexible network architecture and system extension. They provide the two(2) 10/100Mbps fast Ethernet interface for multiple IP Extend Key Terminal interconnection (cascading). Because some user want to use several hundred speed dialing key for dial access speed-up, terminal cascading should be supported.

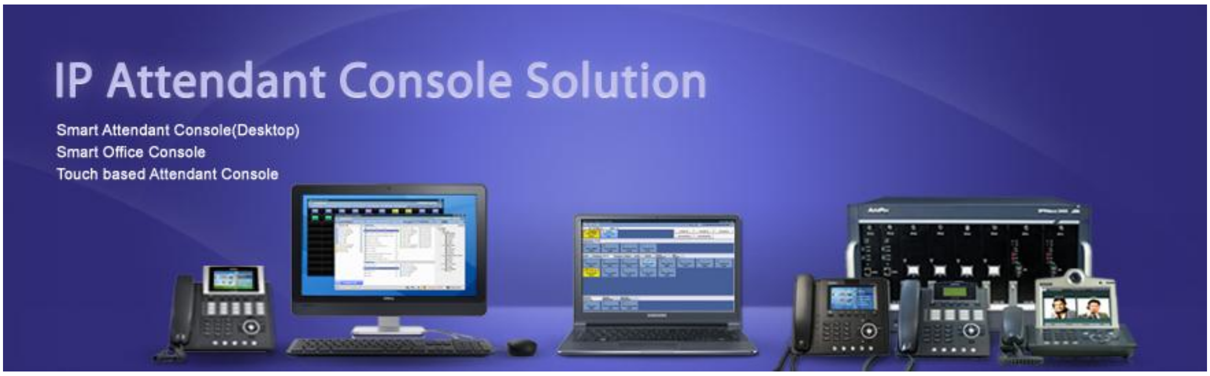
AddPac IP Extend Key Pack Terminal Comparison Table

Model / Service Features	AP-PT100 	AP-PT50 	AP-PT20 
Key Type	7 inch LCD Touch Screen	Push Button with User Presence Indication LAMP	Push Button with User Presence Indication LAMP
Key Number	Default : 9(row) x 4(column) = 36	60 Key	40 Key
User Presence Indication	Support	LED on, LED off, LED Blink	LED on, LED off, LED Blink
Multiple Cascading	Support	Support	Support
Speaker	Support	Support	Support
LAN Port	2	2	2
PoE(Option)	Support	Support	Support
Application	IP Phone or Video Phone Extend Key Pack	IP Phone or Video Phone Extend Key Pack	IP Phone or Video Phone Extend Key Pack

Network Diagram



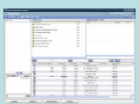
IP Attendant Console Solution



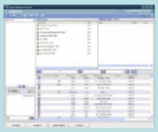
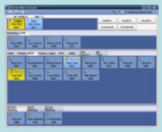
AddPac Technology's Smart Attendant Console is an IP based soft attendant console that interoperates with AddPac IP-PBX products, which are based on its next generation multi-media IP telephony solution. This product together with AP-IP300 IP phone can be used for IP based soft attendant console solution. In large-scale enterprise, government, etc, for inbound/outbound call processing, call transfer, call exchange, dedicated persons or department are necessary. Smart Attendant Console is IP based soft attendant console solution for them. Smart Attendant Console based on MS-Window program supports all different kinds of services such as messenger service, unified message (voice mail, short message), remote IP broadcasting, alarm registration/processing, beside basic attendant console function like as user presence Board, inbound call rerouting, telephone directory. Smart Attendant Console can be operated by AddPac Technology's proprietary protocol called Smart Service Control Protocol (SSCP), which is managed between AddPac's IP PBX and IP terminal like as Video Phone, IP Phone, etc.

AddPac Technology's Smart Office Console is touch screen based light soft attendant console solution for secretary. Smart Office Console based on MS-Window program supports all different kinds of services such as messenger service, unified message (voice mail, short message) beside basic attendant console function like as user presence Board, inbound call rerouting, telephone directory. Smart Office Console can be operated by AddPac Technology's proprietary protocol called Smart Service Control Protocol (SSCP), which is managed between AddPac's IP PBX and IP terminal like as Video Phone, IP Phone, etc.In order to provide User Presence features on real-time basis, such as user busy, on-line, user away, etc, the Presence server can be mounted and operated on IP-PBX. The Presence Server can be operated on the same hardware platform and can be operated independently in the platforms separated to each other.In order to support user presence features on real-time basis, such as user busy, on-line, user away, etc, user presence server function should be provided on IP telephony solution. User Presence Server function gathers user information (on-line, busy, etc) of each IP terminals from IP-PBX, and broadcasts user information to all or group IP terminals with speed-dial keys (built-in presence indication lamp), Smart Messenger Program, Smart Office Console at every second. External presence server such as AP-PS2000 should be used with IP-PBX due to performance issue independently in large scale enterprise for user presence indication.

IP Attendant Console Product Lists

	Smart Attendant Console	Attendant Console Software for Large Company
	Smart Office Console	Attendant Console Software for Secretary (Option : Touch Screen)

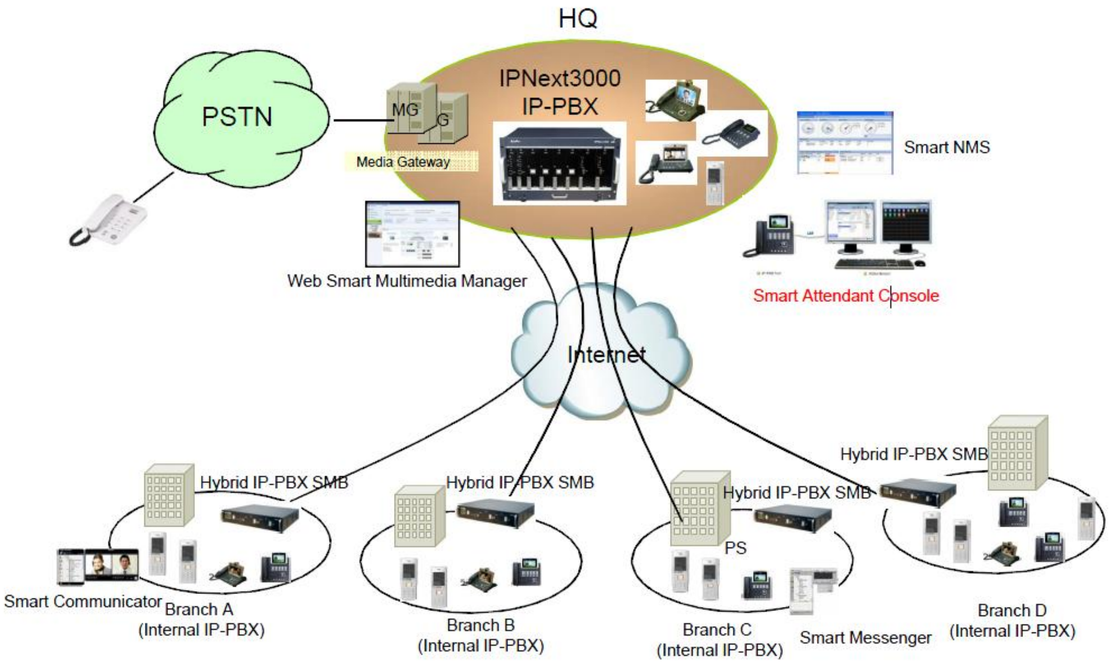
IP Attendant Console Product Comparison

	Smart Attendant Console	Smart Office Console
		
Feature Lists	• Support Call Routing Service • Support Directory Search • Support User Presence Information • Support Phone Number Presence Information • Interoperation with Directory and Smart Attendant Console • Support Smart Call Control and Additional Service Control • Support Longest Wait Time based Incoming Call Process • Support Private Phone Book • Support Drag and Drop Call Control Service • Support Remote Broadcasting Control Feature	• Support Call Routing Service • Support User Presence Information • Support Phone Number Presence Information • Support Attendant Phone Call Control • Support Call Status Display • Support Group Call(Conference) • Support Various Monitoring Group

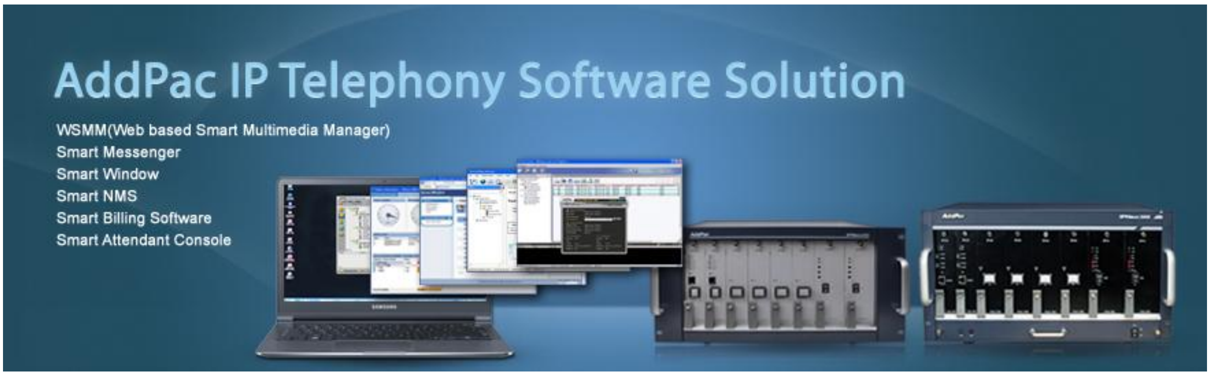
Single Monitor based Smart Attendant Console

Dual Monitor based Smart Attendant Console

Network Diagram

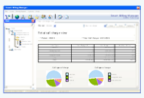
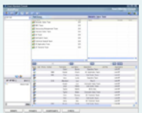


IP Telephony Software Solution



AddPac Technology's IP Telephony Software Solution consists of Web based Multimedia Manager Software for IP-PBX, Network Management System Software for Large Scale Deployment, Smart Messenger for Click-to-Dial, Smart Window for Personal Web Management, Smart Billing Software for Enterprise, Smart Attendant Console Software for Large Company, and Soft Multimedia Phones.

AddPac IP Telephony Software Products

	WSMM	Web based SMM (Smart Multimedia Manager) for IP-PBX
	Smart NMS	Smart NMS (Network Management System)
	Smart Messenger	Smart Messenger for Click-to-Dial
	Smart Window	Smart Window for Personal Web Manager
	Smart Billing Software	Smart Billing Software for Enterprise
	Smart Attendant Console	Attendant Console Software for Large Company
	Smart Communicator	IP Soft Video Phone AP-SMP100

Web based Smart Multimedia Manager (WSMM) is a new management software used for IP-PBX (SIP server features included) MCU devices in the next-generation multimedia telephony solutions of AddPac Technology. WSMM is pure web based management software supporting the user management, device management, various call management, conference call management, event and trace monitoring, and powerful debugging of IP-PBX in various Web Brower environment such as Internet explorer, Smart Phone Web Browser, Tab Web browser.

Smart Network Management System (NMS) is the software that manages IP-PBX (including SIP Server), Unified Messaging Service (UMS) Server, User Presence Server, RBT (RingBack, Coloring) Server, Interactive Voice Recording (IVR) Server, Real-time Transport Protocol (RBT) Proxy Server, Audio/ Video Multipoint Conferencing Unit (MCU), Media Gateways, IP Phones, Video Phone and WiFI Phones systematically. AddPac Smart NMS is configured with Linux Server and web-based Client software and it is the software that supports the features of Network Resource Management, Device Fault Management, Device Fault History Management, Device Status Management, Fault Notification Management, Device Fault Statistics, Device Model & Service Management

Smart Messenger interoperates with AddPac's Customer Premise Equipment (CPE) products. This software can provide all different kinds of services such as Messenger Service , Telephone Directory, User's presence, Unified Massage (Voice Mail, Short Message), Call Control and Forward Setup functions.
Smart Messenger can be operated by AddPac Technology's proprietary protocol called Smart Service Control Protocol (SSCP), which is managed between AddPac's IP PBX and IP CPE's in an environment of Microsoft Window based PC Platform. In order to provide User Presence features on real-time basis, such as user busy, on-line, user away, etc, the Presence server can be mounted and operated on IP-PBX. The Presence Server can be operated on the same hardware platform and can be operated independently in the platforms separated to each other.

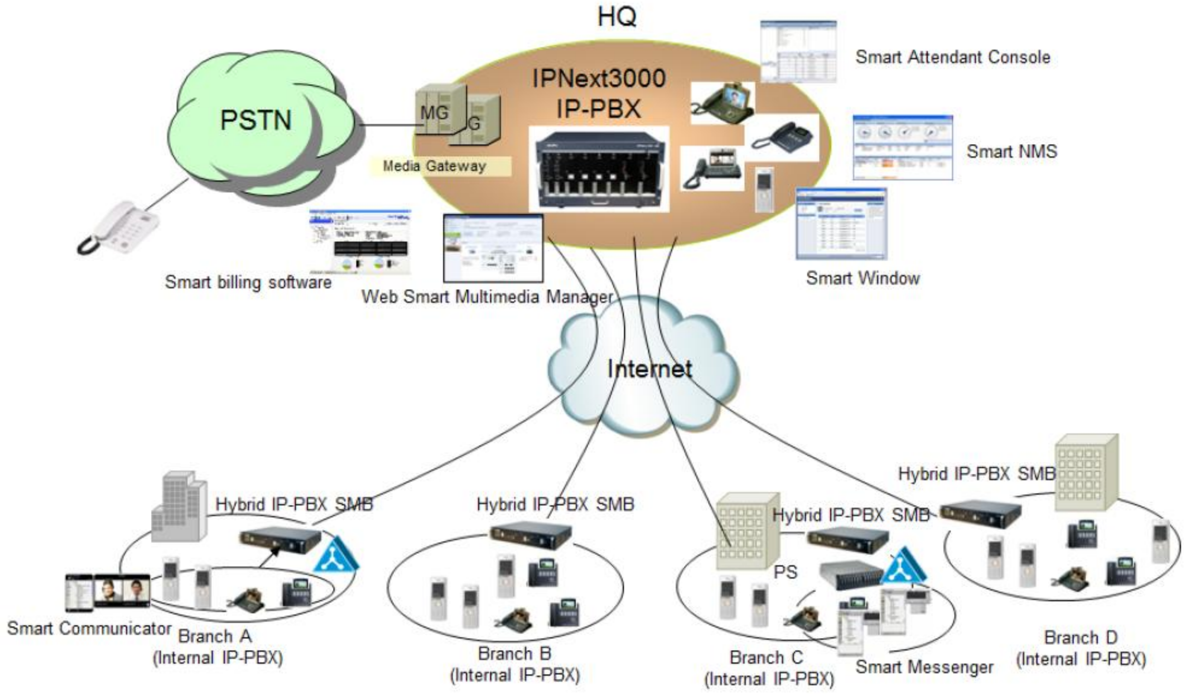
Smart Window helps to setup various private information of IP-PBX registered users using web in multimedia telephony solution. Smart Window provides user information, alarm, search, call forwarding, speed dial, and conference related feature. Enter the login and password to start Smart Window server. The basic configuration screen of Smart Window is composed of menu, contents, and help.

Smart Billing System is a Linux based system to perform efficient communication costs (billing) for enterprise in next generation IP telephony solution. AddPac Smart Billing System is composed of Linux based server and web based client software. It supports tariff management, billing policy management, access level management, system performance management, and billing database backup management.

Smart Attendant Console is an IP based soft attendant console that interoperates with AddPac IP-PBX products, which are based on its next generation multi-media IP telephony solution. This product together with AP-IP300 IP phone can be used for IP based soft attendant console solution. In large-scale enterprise, government, etc, for inbound/outbound call processing, call transfer, call exchange, dedicated persons or department are necessary. Smart Attendant Console is IP based soft attendant console solution for them. Smart Attendant Console based on MS-Window program supports all different kinds of services such as messenger service, unified message (voice mail, short message), remote IP broadcasting, alarm registration/processing, beside basic attendant console function like as user presence Board, inbound call rerouting, telephone directory. Smart Attendant Console can be operated by AddPac Technology's proprietary protocol called Smart Service Control Protocol (SSCP), which is managed between AddPac's IP PBX and IP terminal like as Video Phone, IP Phone, etc.

Smart Communicator AP-SMP100 is a MS Window based Smart Multimedia Soft phone that allows you to make high quality and high resolution video communications by using the Internet. This product supports a video conference, point-to-point video communication, and aid to audio/video communication access for the disabled optimally through the latest audio/video codecs and intelligent smart messenger capability. The criteria of choosing a soft phone are high quality audio/video service and easy dial-access features

Network Diagram



IP Telephony Voice Recording Solution

AddPac Technology IP Telephony Voice Recording Solution provides high performance network based digital voice recording/processing function by interworking with AddPac IP based voice recording device, AddPac VoIP gateway device, IP phone, and IP video phone. AP2650, AP2640, AP2620, AP-MG3000 VoIP gateway, IP phone, and video phone transmits IP packet to the IP based network voice recording device such as AP-NR500, AP-NR700. Scalability/redundancy of system is flexible due to IP based design structure.

IP Based Voice Recording Server Device

Product	AP-NR500 	AP-NR700 	AP-NR2000 
Concurrent Recording User	32channel	64channnel	64channel
Voice Transcoder Module (Option)	Internal module (32channel)	Internal module (64channel)	External (AP-VTC1000)
HDD Module Slot Number(IDE)	One(1)	Two(2)	Ten(10)
LAN Port	2	2	2
Console	1	1	1
Power	Single PSU	Single PSU	Dual PSU

Embedded Voice Recording Service VoIP gateway

Product	AP2620 	AP2640 	AP2650 	AP-MG3000 
Available Modules	AP-FXS4 AP-FXO4 AP-FXS2O2 AP-E&M4 AP-E1	AP-FXS8 AP-FXO8 AP-FXS4O4 AP-E&M8 AP-E1	AP-FXS8 AP-FXO8 AP-FXS4O4 AP-E&M8 AP-E1	APv2-1E1 APv2-2E1 APv2-4E1
Analog Ports	Up to 8	Up to 32	Up to 32	N/A
Signaling	SIP, H.323	SIP, H.323	SIP, H.323	SIP, H.323
Digital E1/T1	Up to 2E1/T1	Up to 2E1/T1	Up to 2E1/T1	Up to 4E1/T1
E&M	Support	Support	Support	N/A
Module Slot	Two(2)	Four(4)	Four(4)	Two(2)
LAN Port	2	2	2	2
Console	1	1	1	1
Power	Single PSU	Single PSU	Dual PSU	Single PSU

Embedded Voice Recording Service IP Phone

	AP-IP300 	AP-IP250 	AP-IP230 	AP-IP160 	AP-IP120 	AP-IP90 
LCD Size	4.3 Inch Color LCD	4.3 Inch Color LCD	5 Inch Color LCD	4 Text Line Graphic LCD	4 Text Line Graphic LCD	4 Text Line Graphic LCD
Touch Screen	N/A	Support	Support	N/A	N/A	N/A
Speed-Dial Keys	25 Key with Presence LED	Touch Screen based 25 Keys	Touch Screen based 25 Keys	16 Key with Presence LED	12 Key with Presence LED	N/A
Voice Codec	G.711/G.726/ G.729/G.723	G.711/G.726/ G.729/G.723	G.711/G.726/ G.729/G.723	G.711/G.726/ G.729/G.723	G.711/G.726/ G.729/G.723	G.711/G.726/ G.729/G.723
Signaling	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP
3-Party Conversation	Support	Support	Support	Support	Support	Support
LAN Port	2	2	2	2	2	2
PoE(Option)	Support	Support	Support	Support	Support	Support
FXO(Option)	Support	Support	Support	Support	Support	Support

Embedded Hybrid IP-PBX (For Branch)

Model / Service Features		IPNext250 	IPNext210 	IPNext190 	IPNext180 
Registration/Concurrent User Number		200/100	150/50	150/50	100/30
IPv4/IPv6 Dual Stack Support		Support	Support	Support	Support
VoIP Signaling	Internal	SIP	SIP	SIP	SIP
	External	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP
Powerful IVR, UMS, Media Service, User Presence Service		Support	Support	Support	Support
RTP Proxy Service (IPv6, Private IP)		Support	Support	Support	Support
LAN Port		2	2	2	2
VoIP Module Slots for PSTN		10 Slots (8 Port Module)	4 Slots (8 Port Module)	4 Slots (8 Port Module)	2Slots (8 Port Module)
VoIP Interface		FXS, FXO,E1/T1	FXS,FXO,E&M, E1/T1	FXS, FXO,E1/T1	FXS,FXO,E1/T1

IP based voice communication is rapidly spread/develop as an alternative solution of existing PSTN telecommunication in accordance with fast internet network BcN (Broadband convergence Network) such as ADSL, VDSL, and FTTH. PSTN network and All-IP based internet VoIP network are coexisted through the development of communication network gradually as broadband IP integrated network. An increase of demand in voice recording solution has been increasing on PSTN network and internet communication network structure. The next generation voice recording solution not only to provide voice recording in PSTN telecommunication network, but it also to provide voice recording in VoIP based internet telecommunication network. AddPac network based voice recording solution is next generation voice recording solution which support total voice recording service architecture in Broadband IP integrated network along with various IP based VoIP devices such as analog/digital VoIP gateway, IP phone, IP video phone, and IP-PBX.

Embedded Type IP Telephony Voice Recording Solution

AddPac VoIP gateway and network voice recording server for voice recording solution provides network based PSTN voice recording server solution by integrating with AddPac high performance CPU module, real time OS (APOS), and embedded hardware.Particularly, AddPac IP based voice recording server device optimizes the performance/efficiency using embedded RISC processor and stabilized embedded OS. AddPac voice recording servers save/manage the compressed voice signal from VoIP gateway as well as transmitting to the media player. Hard disk supports IDE type as well as RAID 1 configuration for hard disk redundancy. Also, it supports Push Button type Hot-Swap to exchange with other hard disk while in service. For backup, it provides network sharing function through secondary storage such as DVD, etc.

Analog and Digital Interface VoIP Gateway

AddPac AP2650 is a midrange VoIP gateway that supports up to 32 ports of analog interface and two digital E1/T1 interfaces. When using AP2650 in enterprise, government office and local government, it may be used as media gateway which interoperates with PSTN PBX and Legacy PBX. It is designed to support high performance service as media gateway in terms of hardware architecture aspects. CPU/network interface module in front of device supports several status inspection LED to check the status of the device at a glance. Most of all, it enhanced the stability of the system using AC power redundancy to cope with power failure issue.

Embedded Voice Recording Service IP Telephone

AddPac IP telephone/video phone supports network based voice recording server device source function. User may record the voice communication by transmitting voice packet from specific IP phone. IP phone user may save the voice information of AP-NR700 in voice recording server by pressing function key if recording is necessary.

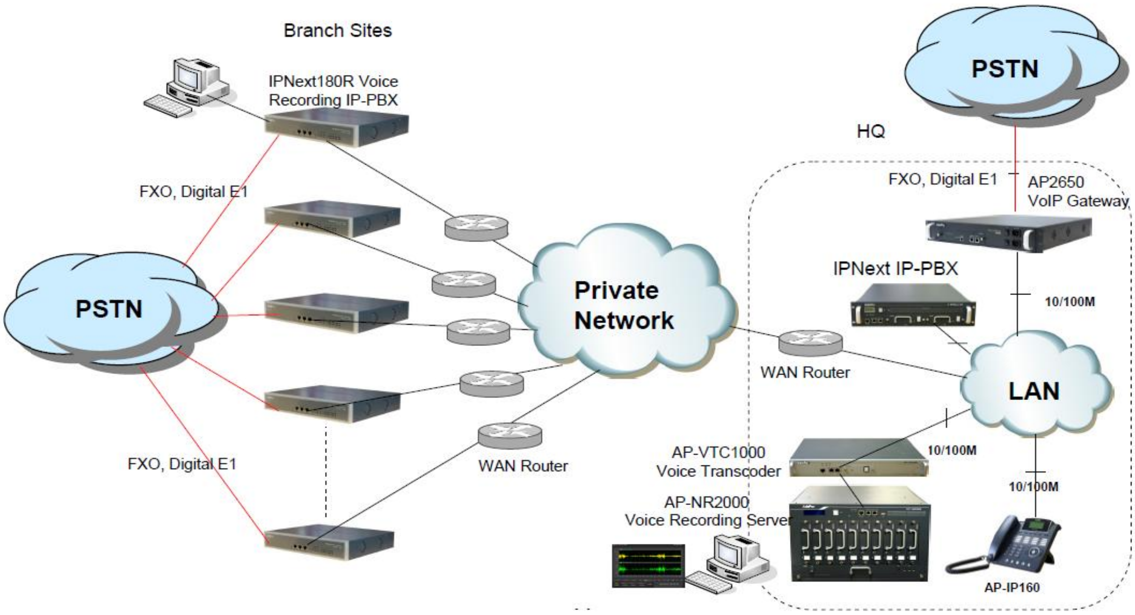
Support High Quality Voice

AddPac IP telephony voice recording solution supports real time voice recording solution using high performance DSP based on an excellent performance and stability. Recorder voice data can be replayed through Window based voice recording exclusive program with speaker or headset.

Firmware Upgradable Hardware Architecture

AddPac IP telephony voice recording solution RISC processor is all programmable structure so that constant feature improvement, modification, and additional feature are possible. User may cope with future technological changes when downloading directly from AddPac homepage or setting an auto upgrade option without much efforts.

Network Diagram



Secure IP Telephony Solution

Various secure IP telephony products (IP-PBX, IP Phone, Video Phone, VoIP Gateway, Video MCU, NMS) of AddPac are approved for its high performance and reliability in world-wide markets. AddPac provides highly advanced IP telephony services in order to meet demanding and evolving business requirements of customers. Especially AddPac Secure IP telephony products have been developed through years of experiences and accumulated know-how from existing enterprises and communication markets. AddPac secure IPNext IP-PBX inter-works with various IP terminals of AddPac (e.g. AP-VP300, AP-VP280, AP-VP250, AP-VP150, AP-VP120 IP Video Phones, and AP-IP300, AP-IP230, AP-IP160, AP-IP120, AP-IP90 IP Phones) using TLS1.x and SRTP secure protocols to provide secure multimedia IP telephony services as well as the traditional legacy PBX features. These IP telephony systems are embedded with encrypt processing function such as AES, SEED and custom encryption algorithm (ARIA, etc) to support SRTP (Secure RTP), TLS 1.x VoIP communication. AddPac secure IP Telephony solution is suitable for medium and large size companies, and inter-works with the VoIP and video products of AddPac Technology such as IP phone, Video Phone, Video MCU, Video Conference System to provide a variety of IP application services appropriate for your network.

Secure Analog VoIP Gateways(32-Port)

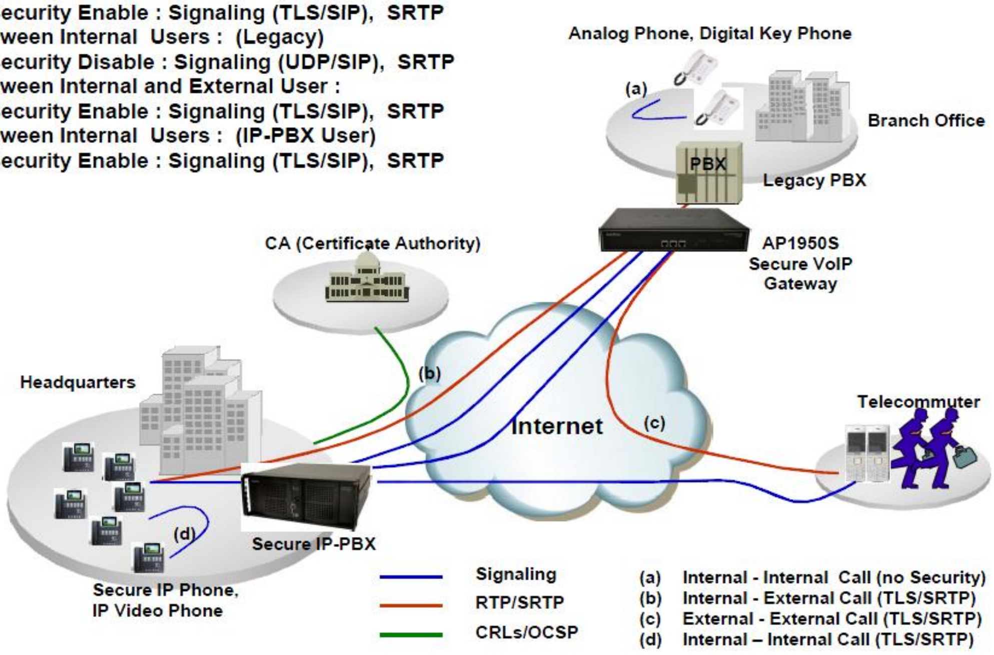
Product	AP2330S 	AP2340S 
Available Modules	AP-N1-FXS8 AP-N1-FXO8 AP-N1-FXS4O4	AP-N1-FXS8 AP-N1-FXO8 AP-N1-FXS4O4
Analog Ports	Up to 24	Up to 32
Signaling	SIP, H.323	SIP, H.323
TLS/SRTP Support	Yes	Yes
Module Slots	3	4
Module Slot	Three(3)	Four(4)
LAN Port	2	2
Console	1	1
Power	Single PSU	Single PSU

Secure Analog VoIP Gateways(32-Port)

Product	AP1900S 	AP1950S 
Available Modules	AP-N1-E1 AP-N1-FXS8 AP-N1-FXO8 AP-N1-FXO8 AP-N1-FXS4O4	AP-N1-E1 AP-N1-2E1 AP-N1-FXS8 AP-N1-FXO8 AP-N1-FXS4O4
VoIP Signaling	SIP, H.323	SIP, H.323
Digital E1/T1	Up to 1E1	Up to 2E1
Digital Signaling	ISDN PRI, R2	ISDN PRI, R2
TLS/SRTP Support	Yes	Yes
Module Slot	Two(2)	Two(2)
LAN Port	2	2
Console	1	1
Power	Single PSU	Single PSU

Network Diagram

- **Between External Users :**
  Security Enable : Signaling (TLS/SIP), SRTP
- **Between Internal Users : (Legacy)**
  Security Disable : Signaling (UDP/SIP), SRTP
- **Between Internal and External User :**
  Security Enable : Signaling (TLS/SIP), SRTP
- **Between Internal Users : (IP-PBX User)**
  Security Enable : Signaling (TLS/SIP), SRTP



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