AddPac
VoIP Solution

VoIP Solution

IP Audio Broadcasting Solution

AddPac Technology IP Audio Broadcasting solution is composed of AP-ABS5000 multi-channel IP audio broadcasting server, AP-BCR2000 Broadcasting Router, AP1605 high quality IP audio broadcasting terminals, and AP-MBMS broadcasting management program. AP-ABS5000 multi-channel IP audio broadcasting server is an internet access ready IP broadcasting device to support multi-channel audio and voice broadcasting transmission by using IP protocol in finance, enterprise, and public office. It supports high definition audio quality up to 16KHz with high quality audio compressed codec such as MP3, WMA, and AAL-LC. AP-ABS5000 multi-channel IP audio broadcasting server supports voice broadcasting through internet in enterprise environment as well as excellent solution in video broadcasting to satisfy the needs of customer demand. Beside, VoiceFinder AP-ABS5000 IP audio broadcasting server supports stabilized two fast Ethernet service port and let you install maximum of 10 audio codec modules. AP1605 high quality IP audio broadcasting terminals is a fast internet IP broadcasting terminal which provide audio, voice broadcasting data play through internet IP protocol in headquarter and branch. It provides high definition audio quality (up to 16KHz) by using high quality audio compressed codec such as MP3, WMA, AAL-LC other than existing 8KHz based PSTN. At front, it provides volume adjust rotary switch so that a user may adjust volume in accordance with the needs of user. Also, it provides built-in digital amp broadcasting module with direct connection of external speaker. AP1605 IP audio broadcasting terminal supports an excellent solution for voice and broadcasting task through internet in enterprise environment to maximize the full satisfaction. AP1605 IP broadcasting terminal is designed on module type to support audio broadcasting interface module such as built-in amp. AP-BCR2000 broadcasting router is a IP based broadcasting router which is operated along with IP broadcasting server to broadcast the compressed IP audio bit stream in form of 1:N for broadcasting or multicasting broadcast. At the front, it supports LED to display 2 port 10/100/1000Mbps Gigabit Ethernet interface, RS-232 console interface for CLI (Command Line Interface), CPU and network utilization in real time. It is applicable to various IP based real time audio broadcasting fields along with IP audio broadcasting server and IP audio broadcasting terminal such as audio/voice band broadcasting, announcement, etc

IP HQ Audio Broadcasting Solution Product Table

Audio Broadcasting Manager S/W	HQ Audio Broadcasting Server AP-ABS5000	Audio Broadcasting Router (Relay Server)	HQ Audio Broadcasting Terminal AP1605A
			
Window based Audio Broadcasting Management Software.	Embedded Hardware based Audio Codec. Ten(10) HQ Audio Codec Module. MP3, AAC-LC, WMA,G.711 Audio Codec.	1:N Audio Broadcasting Router. Gigabit Ethernet Support	Embedded Hardware based Audio Terminal. Volume Control Rotary Switch. One(1) HQ Audio Codec Module. Built-in AMP. MP3, AAC-LC, WMA,G.711 Audio Codec.

AP-ABS5000 Audio Module

Audio Module Type (AP-N1-HQA1000)	Audio Module Features	Maximum Audio Channel in AP-ABS5000
	1-Channel Audio In/Out Port	Up to 10 channel = 10 Module x 1 Channel
	Audio IN : MIC, Line IN Audio OUT : Headphone, Line OUT 3.5mm Stereo JACK	
	High Quality MP3,WMA,AAC-LC,G.711 Audio Codec	

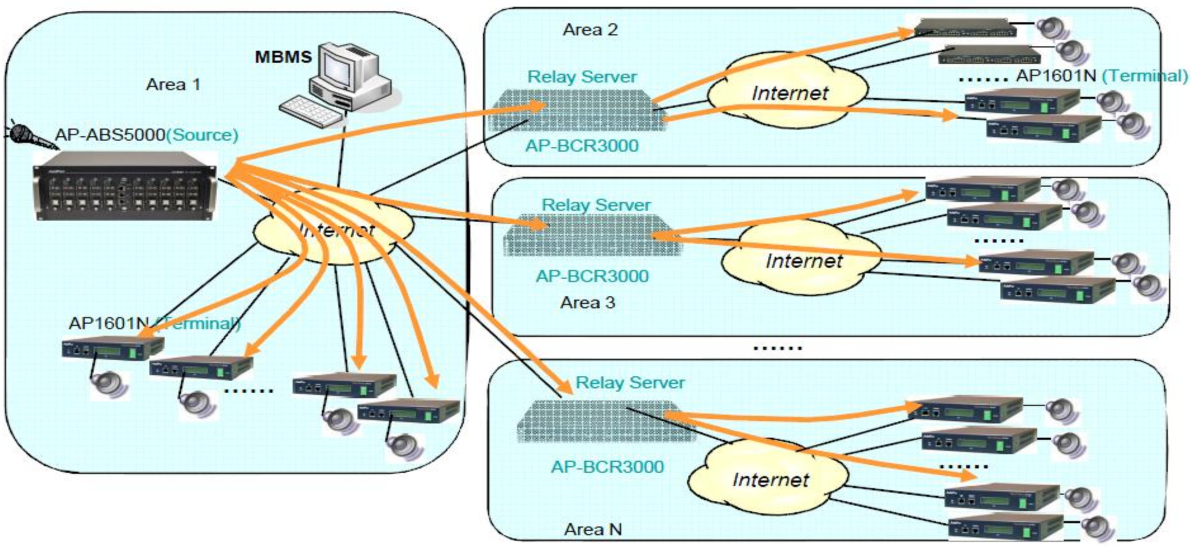
AP-ABS1605 Audio Module

Audio Module Type	Audio Module Features
AP-N1-HQA1000A 	One(1)-Channel Audio In/Out Port
	One(1) Fast Ethernet Port
	One(1) RS232C Port
	Audio Encoding/Decoding Service
	Audio IN Audio OUT AMP. Built-in Speaker Out (Left, Right) 4ohm Speaker : 50Watt 8ohm Speaker : 30Watt
	High Quality MP3,WMA,AAC-LC,G.711 Audio Codec

Audio Module Type	Audio Module Features
AP-N1-HQA1000 	One(1)-Channel Audio In/Out Port
	One(1) Fast Ethernet Port
	One(1) RS232C Port
	Audio Encoding/Decoding Service
	Audio IN : MIC, Line IN Audio OUT : Headphone, Line OUT 3.5mm Stereo JACK
	High Quality MP3,WMA,AAC-LC,G.711 Audio Codec

Network Diagram

Unicast Method



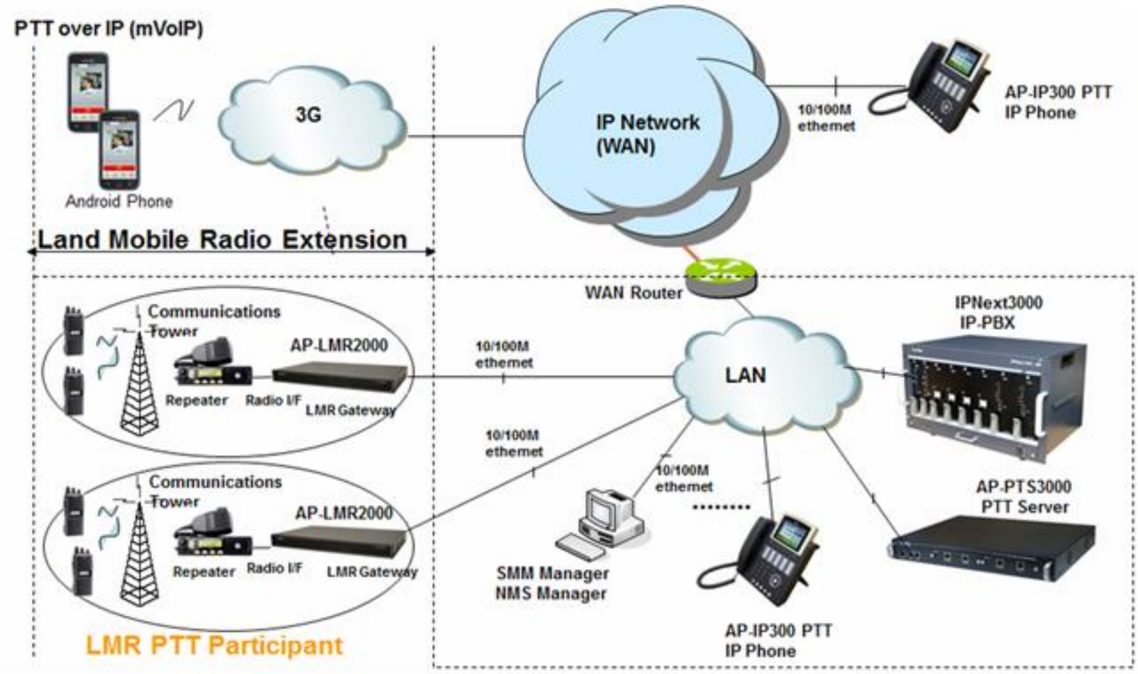
PTT over IP Solution

AddPac Technology PTT over IP solution is a PTT group call service total solution which can be used under various wire/wireless environments such as 3G smart phone (mVoIP) and existing radio device network. SIP based AddPac group call solution is composed of IPNext3000, AP-PTS3000PTT server, LMR(Land-to-Mobile Radio) Gateways, WiFi IP Phone, Android based smart phone SIP PTT application, PTT IP phone, and network management system NMS (Network Management System). AddPac group call solution is designed on the basis of IP network to support 1:1 and group call service in any places. It allows saving communication costs with VoIP technology. The main advantage of this device is that it supports radio device network function so that user can maintain existing radio device network system and merits. It also supports group call service by expanding the radio device network coverage to nationwide 3G network. It is designed to expand wire/wireless PTT service users by increasing the number of PTT server. Particularly, Android smart application supports 3G (mVoIP), and Wifi dual mode. It uses the verified standard signal system so that it has both excellent compatibility and scalability.

PTT over IP Solution Product Table

IP-PBX	IPNext3000	
PTT Server	AP-PTS3000	
LMR Gateway	AP-LMR2000	
PTT Media Gateway	AP2650PMG	
PTT GSM Gateway	AP-GS3500PGG	
PTT IP Phone	AP-IP300, AP-IP230	
PTT WiFi Phone	AP-WP100	
PTT Smart Phone Appl.	Android (AP-SAP100) I-Phone (AP-SIP100)	

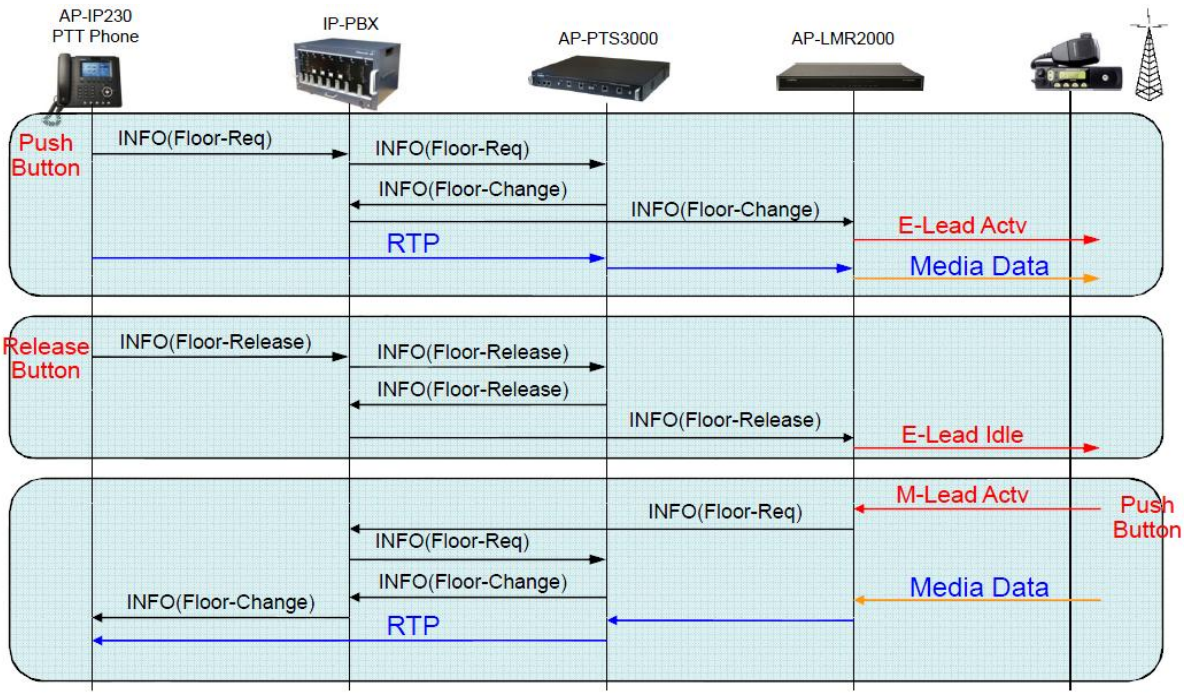
Network Diagram



LMR(Land-to-Mobile Radio) Gateway Solution

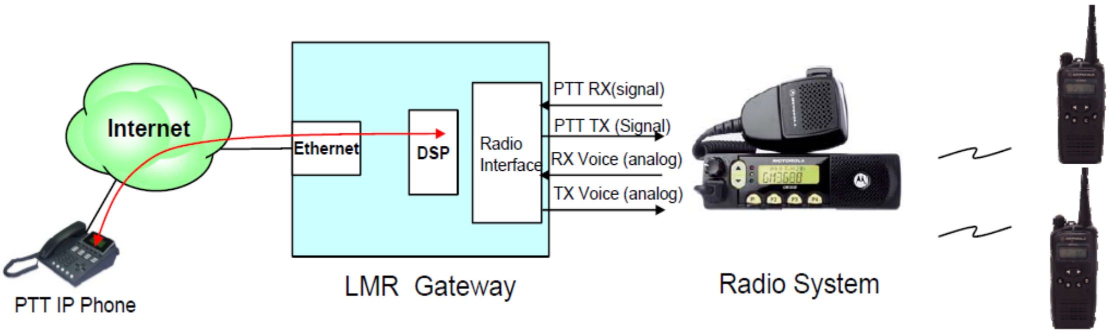
AddPac Technology Radio over IP solution is a PTT group call service total solution which can be used under various wire/wireless environments such as 3G smart phone (mVoIP) and existing radio device network. Especially, AddPac LMR VoIP Gateway supports Radio over IP service for existing Radio Device like as Motorola Radio System. (VoIP?E&M style Radio Interface->Radio System). SIP based RoIP PTT group call solution is composed of IPNext3000, AP-PTS3000 PTT server, LMR(Land-to-Mobile Radio) Gateways, PTT IP phone, and network management system NMS (Network Management System).

RoIP System Message Flow



AddPac RoIP PTT group call solution is designed on the basis of IP network to support 1:1 and group call service in any places. The main advantage of this device is that it supports radio device network function so that user can maintain existing radio device network system and merits. It also supports PTT group call service by expanding the radio device network coverage to worldwide via internet by using VoIP Technology.

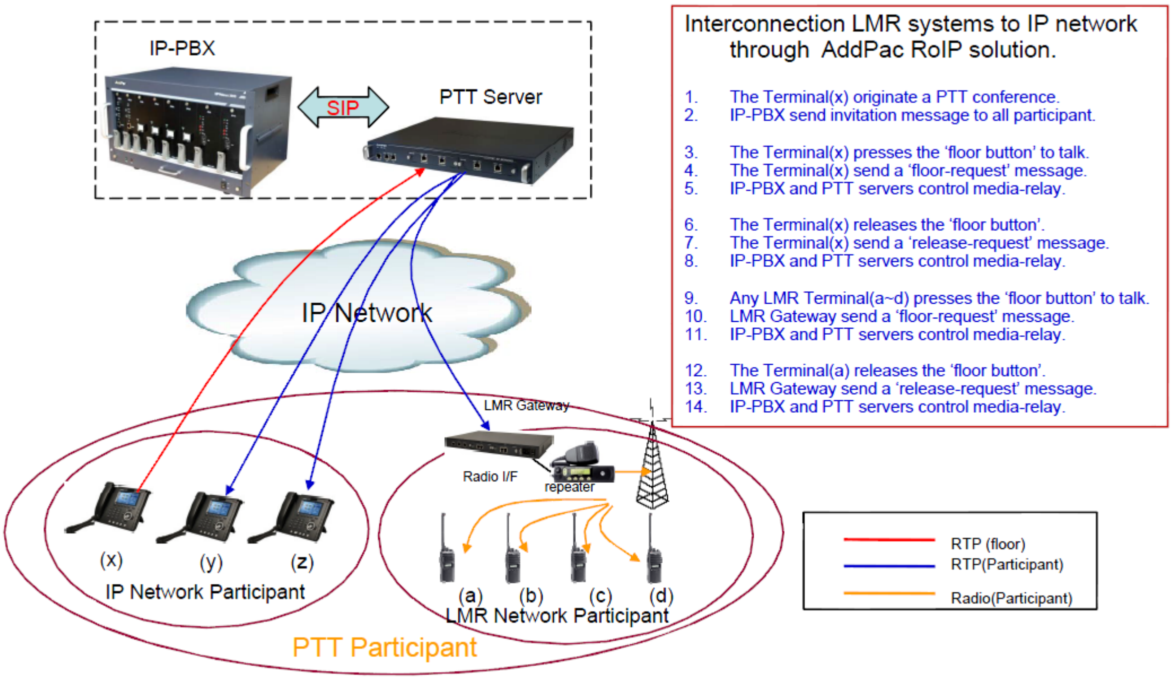
System Interface between LMR Gateway and Radio System



LMR Gateway Comparison Table

Model	AP-LMS1000 	AP-LMS2000 
Available Modules	AP-RADIO2 E&M	AP-RADIO2 E&M
Radio Channel	Up to 2 Ch.	Up to 4 Ch.
Module Slot for Radio Interface	One(1) Module Slots	Two(2) Module Slots
LAN Port	2	2
Console	1	1
Power	Single PSU	Single PSU

Network Diagram



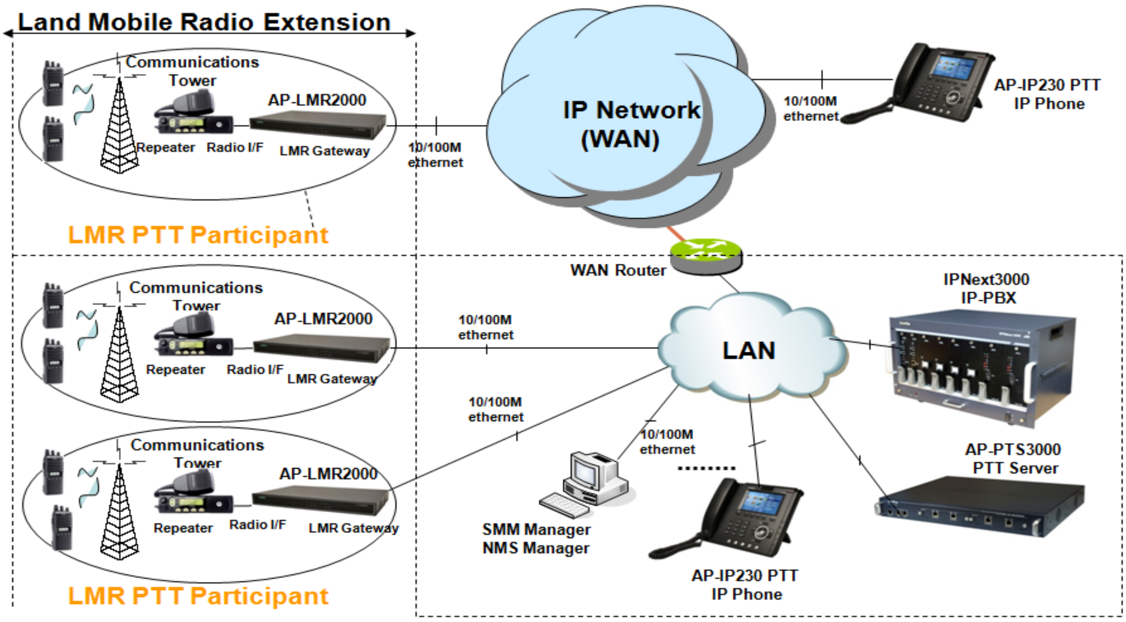
Radio Over IP Solution

AddPac Technology Radio over IP solution is an IP extension PTT group call service solution for radio network environments such as existing radio device network, for example, Motorola radio, etc. The standard SIP VoIP signaling protocol based AddPac RoIP PTT group call solution is composed of IPNext2000, AP-PTS3000 PTT server, LMR(Land-to-Mobile Radio) gateways for radio extension, PTT IP phone, and network management system NMS (Network Management System). AddPac RoIP group call solution is designed on the basis of IP network to support 1:1 and PTT group call service in any places. It allows saving communication costs with VoIP technology. The main advantage of this device is that it supports radio device network function so that user can maintain existing radio device network system and merits. It also supports group call service by expanding the radio device network coverage to worldwide VoIP network. It is designed to expand wire/wireless PTT service users by increasing the number of PTT server. It uses the verified standard VoIP signaling system so that it has both excellent compatibility and scalability.

RoIP Solution Component List

IP-PBX	IPNext2000	
PTT Server	AP-PTS3000	
LMR Gateway	AP-LMR2000	
PTT IP Phone	AP-IP300, AP-IP230	

Network Diagram



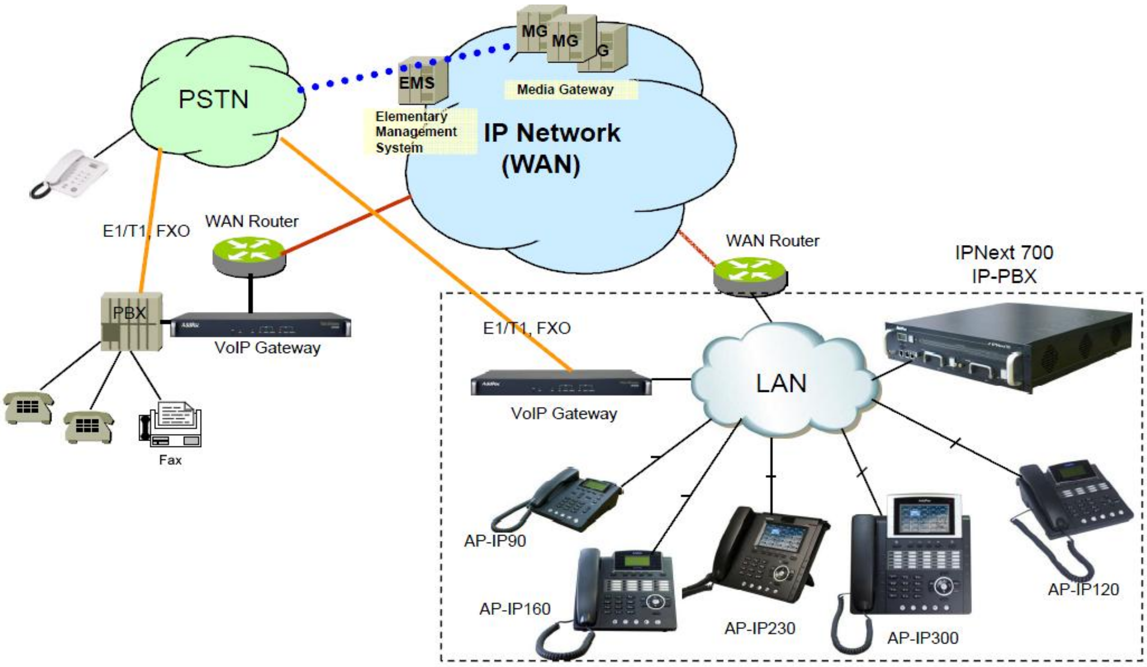
IP Phone Solution

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

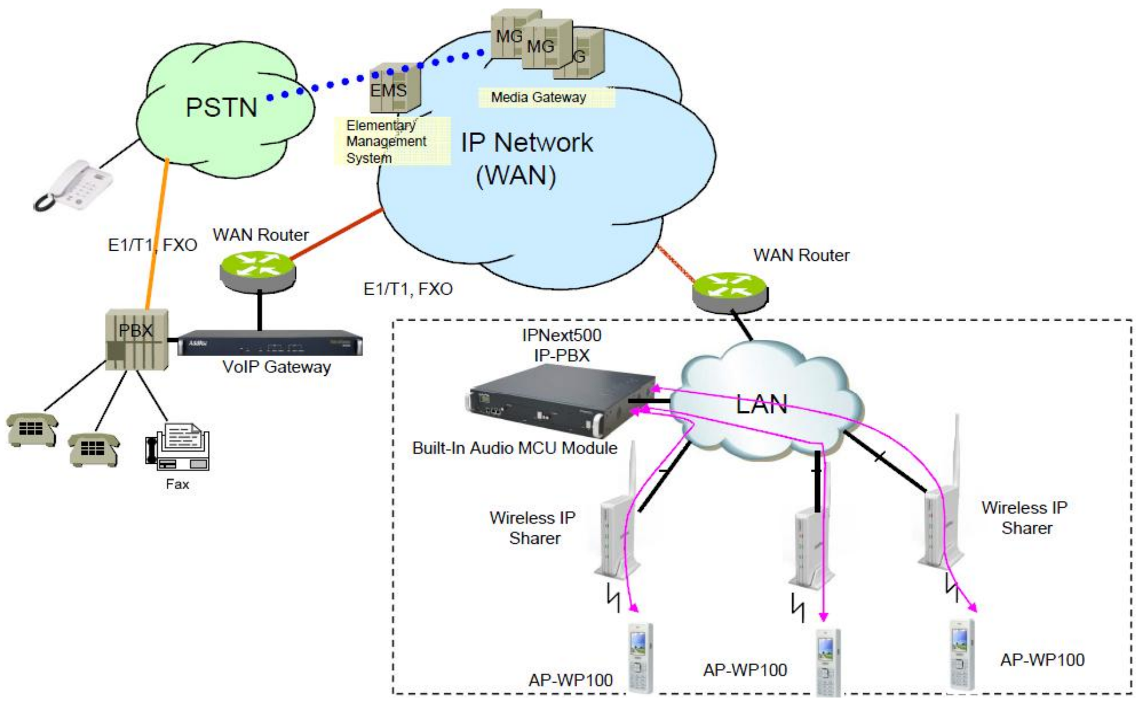
Network Diagram



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 Audio Conference Solution

AddPac Technology IP audio conference solution is composed of AP-MC1500 IP audio MCU(multipoint conferencing unit), fault tolerant IP-PBXs, IP phones, VoIP gateways, and SMM smart multimedia management program. IP based audio conferencing solution of AddPac Technology is a central voice processing solution which provides a wide variety of features such as easy and high level conference control, user-friendly management, powerful audio decoding/mixing/encoding, transcoding and switching capability through reliable real-time encoding and decoding of transmitted audio data from multiple IP audio endpoints such as IP Phone, VoIP Gateway. AP-MC1500 is a hardware type, state-of-the-art equipment for multipoint audio conferencing and real-time collaborative IP communications over internet. AP-MC1500 provides a complete solution specifically for large audio conferences with multiple sites participating which are implemented technically by connecting mixed audio data stream from independent terminals to single 'virtual group' for central mixing and transcoding process. Designed on the basis of firmware upgradeable high performance DSP, AP-MC1500 supports not only latest audio codec currently but has capabilities of new codec services leveraging continuous firmware level upgrade. AP-MC1500 is also a key component of a comprehensive solution combined with AddPac's IP-PBX (especially large capacity, IPNext3000, IPNext200, etc), IP phone equipments such as AP-IP300 IP phones, various VoIP gateways (1~256 Port analog VoIP gateway) for multipoint audio conferencing, which provides easy-to-use user interfaces, robust performance and excellent audio quality. AP-MC1500 MCU system for multipoint audio conferencing features two(2) Fast Ethernet interfaces, a RS-232C console port for maintenance and two(2) hardware module slots for optional card. Hardware based MCU module (HIM-AMCU128, HIM-AMCU64, etc) and diverse network interface (V.35, ATM, POS) modules can be equipped in AP-MC1500's module slot interface.
Designed on the foundation of parallel DSP architecture for real-time processing, MCU module provides a full suite of functionality for rich multipoint audio conferencing experience including a wide range of transcoding coverage from voice band G.711, G.726, G.729, G.723.1, etc audio signals.

IP-PBX Comparison Table

Model / Service Features		IPNext3000 	IPNext2000 	IPNext600 
Registration User Number		3000	2000	500
Concurrent Call User Number		800	500	100
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
System Duplication		Support(built-in)	Support(built-in)	Support(Built-In)

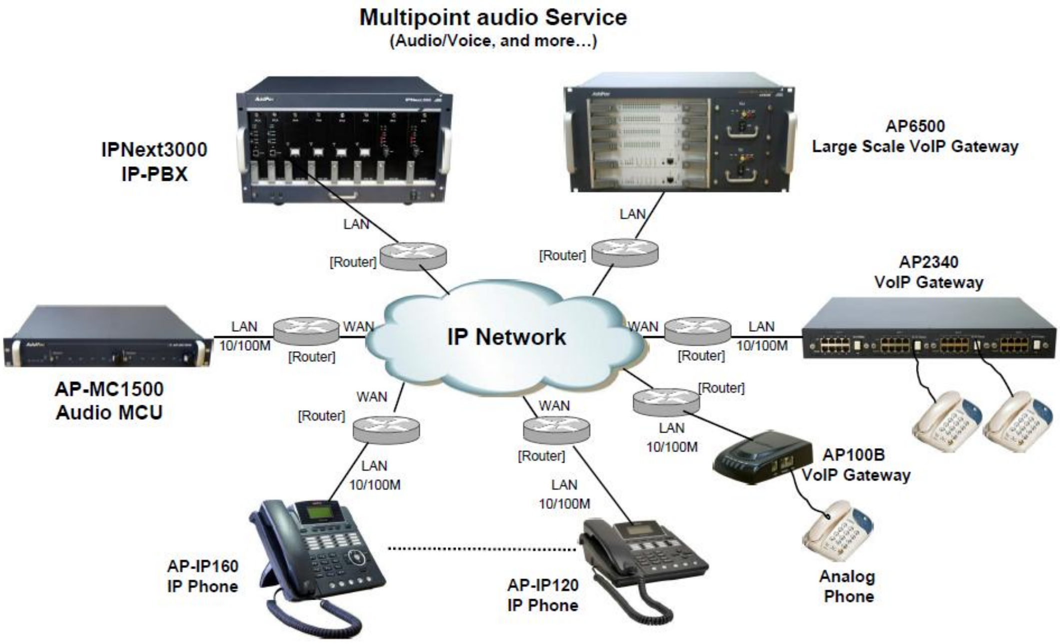
Large Capacity VoIP Gateway Comparison Table

Product	AP3100P 	AP6200 	AP6500 	AP6800 
Available VoIP Modules	AP-FXS4, AP-FXO4 AP-FXS2O2, AP-E&M4	AP-N1-FXS8, AP-N1-FXO8, AP-N1-E&M4, AP-N1-FXS4O4, AP-N1-E1/T1, AP-N1-2E1/T1	AP-N1-FXS32 AP-N1-FXO32	AP-N1-FXS32 AP-N1-FXO32
Analog Ports	Up to 60 (4-Port Module x 15)	Up to 80 (8-Port Module x 10)	Up to 128 (32-Port Module x 4)	Up to 256 (32-Port Module x 8)
Signaling	SIP, H.323	SIP, H.323	SIP, H.323	SIP, H.323
CPU Redundancy (Dual CPU)	N/A	N/A	Support (Option)	Support (Option)
E&M	Support	Support	N/A	N/A
Module Slot for VoIP Module	15 Slots	10 Slots	4 Slots	8 Slots
LAN Port	2	2	2	2
Console	1	1	1	1
Dual Power Supply (Option)	Support	Support	Support	Support

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



PSTN Voice Recording Solution

AddPac PSTN voice recording solution is composed of AddPac VoIP gateway device along with AddPac IP based voice recording device to provide high performance based digital voice recording and processing function. AP2650, AP2640, AP2620, and AP-MG3000 VoIP gateway are connected in between PSTN and existing Legacy PBX to convert into the IPbased voice signal and transmit to IP based network voice recording devices such as AP-NR500, AP-NR700. It offers a better voice quality compare to the existing bridge tap method solution. A system scalability and redundancy are possible due to IP based system.

IP Based Voice Recording Servers

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

IP based voice communication is growing fast as a substitution of existing PSTN phone network in accordance with fast network BcN (Broadband Convergence Network) such as ADSL, VDSL, and FTTH. The demand for voice recording solution has been increasing in the structure of PSTN network and internet network. To meet the needs of customer satisfaction, next generation voice solution should be able to fulfill voice recording in PSTN network as well as in VoIP (Voice over IP) based internet network. AddPac network based voice recording solution supports not only voice recording solution in one access point but it also supports total voice recording service architecture in BcN (Broadband convergence Network) along with various IP based VoIP devices such as analog/digital VoIP gateway, IP phones, and IP-PBX. IP based PSTN voice recording solution would satisfy the needs of demand.

Embedded Type PSTN Voice Recording Solution\

AddPac voice recording VoIP gateway and network voice recording server offers network based PSTN voice recording server solution with high performance CPU module, real-time OS (APOS) and embedded hardware. Especially, AddPac IP based voice recording server device optimizes the performance and efficiency with a high performance embedded RISC processor which support Giga bit Ethernet as well as stable embedded OS. AddPac voice recording server conducts save/manage the compressed voice signal from various VoIP gateways and transmit to the media player for play. Hard disk supports IDE type and RAID 1 configuration for hard disk redundancy. Also, it supports Push-Button type Hot-Swap function for easy hard disk change.

Support Analog and Digital PSTN Interface

AddPac AP2650 VoIP gateway is a midrange VoIP gateway which supports 32 analog interface ports along with two digital E1/T1 interfaces. If enterprise and public office applies AP2650 as voice recording device, it would serve as media gateway which interworks with PSTN PBX and Legacy PBX. AP2650 is designed to provide a high performance service as a media gateway. CPU and network interface module supports several status check LED to check the device status and support AC power redundancy for increasing stability.

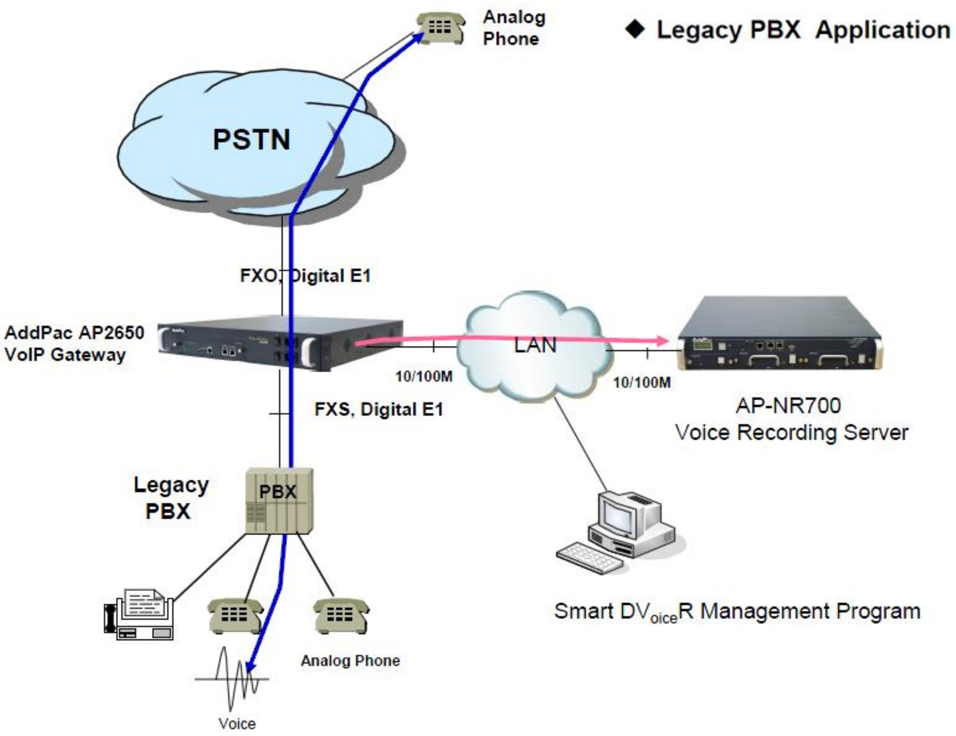
Support High Quality Voice Mechanism

AddPac PSTN voice recording solution provides real time high quality voice recording solution based on DSP, high performance and stability. A user may listen to the recorded voice with speaker headset from AddPac window based vpoce recording exclusive program.

Firmware Upgradeable Hardware Architecture

AddPac PSTN voice recording solution's high performance RISC module is a programmable structure so that it is capable of improving functions, altering, and adding new features constantly. When additional features or changes are needed, simply download from the homepage or setting an auto upgrade option. You may use the latest features without much effort.

Network Diagram



GSM Voice Recording Solution

AddPac GSM voice recording solution is composed of AddPac GSM gateway device along with IP based voice recording device to provide high performance based digital voice recording and processing function. AP-GS3000, AP-GS2800, and AP-GS2500 GSM gateway are connected in between GSM network and existing Legacy PBX to convert into the IP based voice signal, and transmit to IP based network voice recording server such as AP-NR500, AP-NR700. It offers a better voice quality compare to the existing bridge tap method solution. A system scalability and redundancy are possible due to IP based system. The demand for GSM voice recording solution has been increasing in the structure of GSM network and internet network. To meet the needs of customer satisfaction, next generation voice solution should be able to fulfill voice recording in GSM network as well as in VoIP (Voice over IP) based internet network. AddPac IP based voice recording solution supports not only voice recording solution in one access point but it also supports total voice recording service architecture in BcN (Broadband convergence Network) along with various VoIP devices such as GSM gateway, GSM VoIP gateway, analog/digital VoIP gateway, IP phones, and IP-PBX. IP based GSM voice recording solution would satisfy the needs of demand.

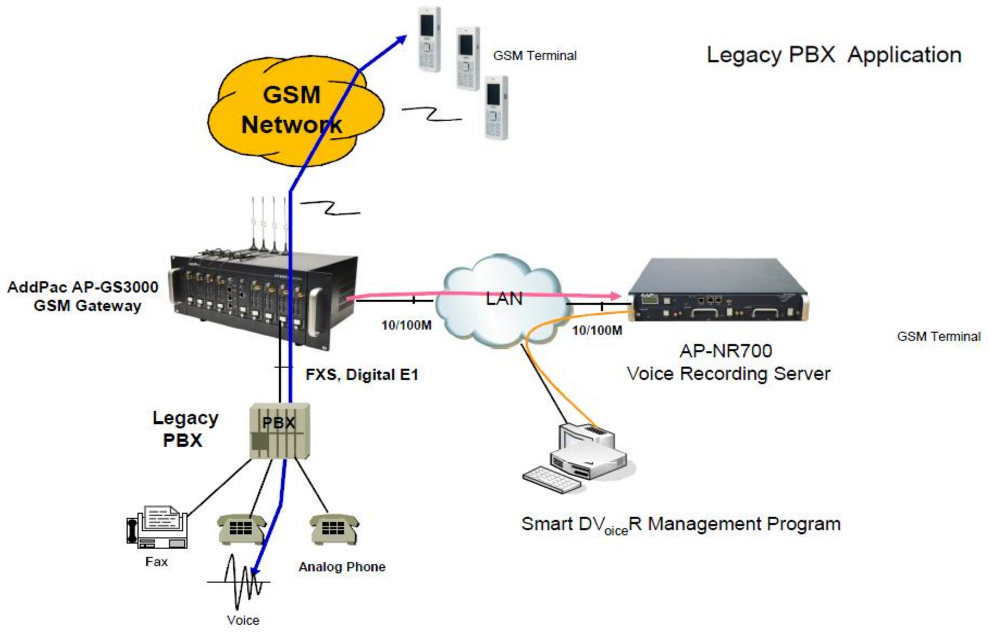
IP based Voice Recording Servers

Product	AP-NR500 	AP-NR700 	AP-NR2000 	AP-NR3000 
Concurrent Recording User	32channel	64channnel	64channel	256channel
Voice Transcoder Module(Option)	Internal module (32channel)	Internal module (64channel)	External (AP-VTC1000)	External (AP-VTC1000)
HDD Module Slot Number	One(1) : IDE	Two(2) : IDE	Ten(10) : IDE	Four(4) : SATA
LAN Port	2 (Fast Ethernet)	2 (Fast Ethernet)	2 (Fast Ethernet)	1 (Gigabit)
Console	1	1	1	N/A
Power	Single PSU	Single PSU	Dual PSU	Single PSU

GSM Gateways for Voice Recording Extension

Model	AP-GS2500 	AP-GS2800 	AP-GS3000 
Available Modules	AP-N1-GSM4 AP-N1-FXS8 AP-N1-FXO8 AP-N1-FXS4O4 AP-N1-E1	AP-N1-GSM4 AP-N1-FXS8 AP-N1-FXO8 AP-N1-FXS4O4 AP-N1-E1	AP-N1-GSM4 AP-N1-FXS8 AP-N1-FXO8 AP-N1-FXS4O4 AP-N1-E1
GSM Channel	Up to 16 Ch.	Up to 28 Ch.	Up to 36 Channel
GSM Antenna	One(1) / 4 Channel GSM Module (AP-N1-GSM4)	One(1) / 4 Channel GSM Module (AP-N1-GSM4)	One(1) / 4 Channel GSM Module (AP-N1-GSM4)
Module Slot	Four(4) Module Slots for GSM	Seven(7) Module Slots for GSM, E1/T1 Module Slot	Nine(9) Module Slots for GSM, E1/T1 Module Slot
LAN Port	2	2	2
Console	1	1	1
Power	Single PSU	Single PSU	Single PSU

Network Diagram



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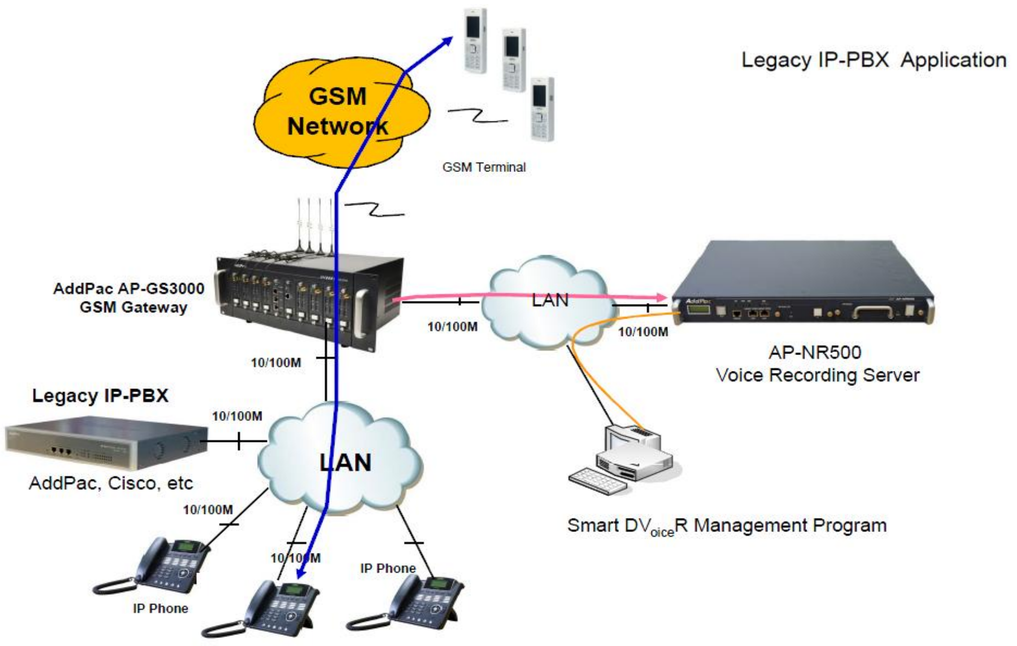
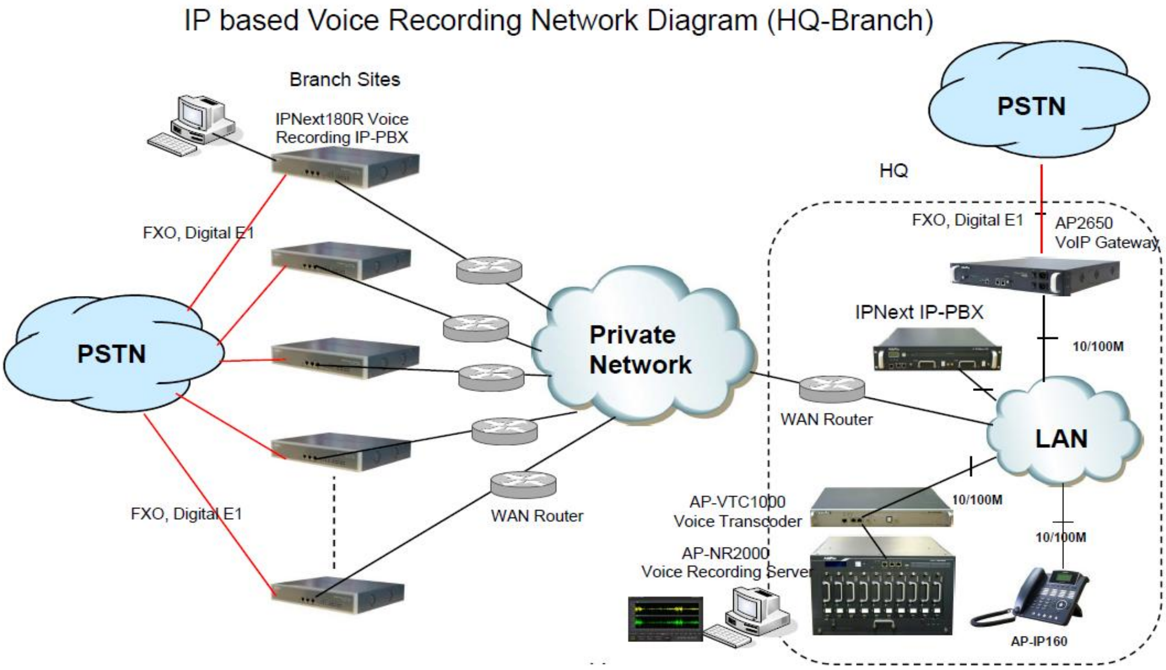


<Legacy IP-PBX Application>

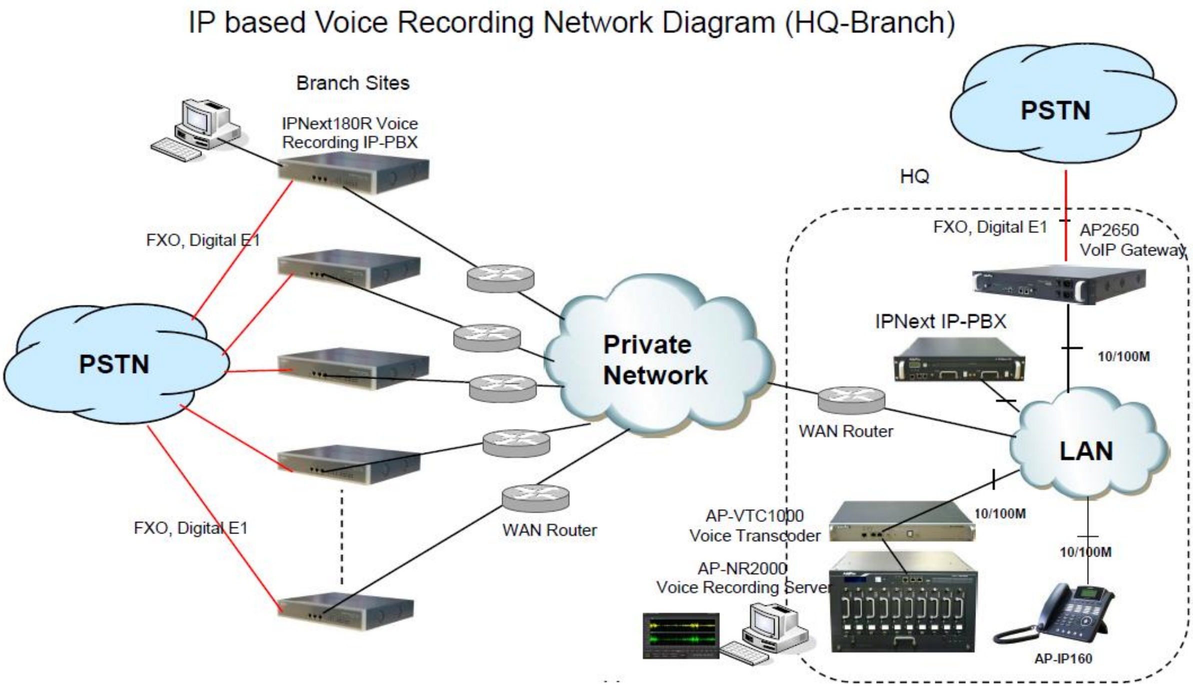
IP Voice Transcording Solution

AddPac IP voice transcoding device AP-VTC1000 supports network based voice codec transcode service with high performance CPU module, real time OS (APOS), and embedded hardware. It has both a high performance RISC processor which support Gigabit Ethernet and stabilized embedded real-time OS to optimize the performance and efficiency. This is a IP-to-IP voice transcoding device for voice recording server (without voice codec transcode function) and convert G.723, G729, G726 into G.711. A voice recording server performs to save/manage the voice signal from various end-point based on G.711 and helps to transmit to media player. AddPac AP-VTC1000 IP voice transcoding device supports a real time high quality voice codec transcode solution based on the state-of-the-art technology DSP. A captured voice data from end-point (VoIP gateway, IP Phone, etc) will be transmitted to commercial based recording server through IP network after voice transcoding. It not only provides voice codec such as G.711, G.726, but also low bit rate VoIP such as G.729, G.723.1. Transcoder module supports up to 64 channels with PCM (G.711).

Network Diagram



<Voice Transcoding Application for VoIP Codec>

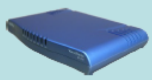
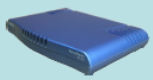


<Voice Transcoding Application for Voice Recording>

VoIP Gateway Solution

Various VoIP gateway series of AddPac is approved for its high performance and reliability in world-wide markets. AddPac provides highly advanced VoIP services in order to meet demanding and evolving business requirements of customers. Especially AddPac VoIP Gateway has been developed through years of experiences and accumulated know-how from existing enterprises and communication markets. AddPac VoIP Gateway provides concurrent triple stack such as H323, SIP, MGCP and QoS (Quality of Service) for high quality of communication service. This will satisfy customers with high expectation.

VoIP Gateway for SOHO (1~2Port)

Product	AP100 		AP100B 		AP190 	AP200 		AP250 	
Model	Type	VoIP	Type	VoIP		Type	VoIP	Type	VoIP
	P	PSTN Backup	B	2FXS, PSTN Backup		B	2FXS, PSTN Backup	B	2FXS, PSTN Backup
						D	2FXO	D	2FXO
						E	1FXS, 1FXO		
VoIP Ports	1		2		1	2		2	
Signaling	SIP, H.323		SIP, H.323		SIP,H.323	SIP, H.323		SIP, H.323	
Module Slot	N/A		N/A		N/A	N/A		N/A	
LAN Port	2		2		2	2		2	
Console	N/A		N/A		Support	Support		Support	
Power	External Adaptor		External Adaptor		External Adaptor	External Adaptor		External Adaptor	

VoIP Gateway for SMB (4Port)

Product	AP700 		AP1000 	AP1002 	AP1005 	AP800 	
Model	Type	VoIP				Type	VoIP
	P	1-Port PSTN Backup				A	1-Port PSTN Backup
						B	4-Port PSTN Backup
VoIP Ports	4-Port FXS		4-Port FXS	2-Port FXS & 2-Port FXO	4-Port FXO	4-Port FXS	
Signaling	SIP, H.323		SIP, H.323	SIP,H.323	SIP, H.323	SIP, H.323	
Module Slot	N/A		N/A	N/A	N/A	N/A	
LAN Port	2		2	2	2	2	
Console	Support		Support	Support	Support	Support	
Power	External Adaptor		External Adaptor	External Adaptor	External Adaptor	External Adaptor	

VoIP Gateway for MB (8Port)

Product	AP900 		AP1100 		AP1200 	
Model	Type	VoIP	Type	VoIP	Type	VoIP
	A	4 FXS, 4 FXO	A	4 FXS, 4 FXO	A	4 FXS, 1 FXO
	B	8 FXS	B	8 FXS	B	8 FXS, 1 FXO
	C	8 FXO	C	8 FXO		
VoIP Ports	Up to 8-Port		Up to 8-Port		Up to 8-Port	
Signaling	SIP, H.323		SIP, H.323		SIP, H.323	
Module Slot	N/A		N/A		N/A	
LAN Port	2		2		2	
Console	Support		Support		Support	
Power	External Adaptor		External Adaptor		External Adaptor	

VoIP Gateway (~24Port Analog, 2E1/T1)

Product	AP1700 	AP1800 	AP2610 	AP2620 	AP2120N 	AP2330 
Available Modules	AP-FXS4 AP-FXO4 AP-FXS2O2 AP-E&M4 AP-E1	AP-N1-FXS8 AP-N1-FXO8 AP-N1-FXS4O4 AP-N1-E1	AP-FXS4 AP-FXO4 AP-FXS2O2 AP-E&M4	AP-FXS4 AP-FXO4 AP-FXS2O2 AP-E&M4 AP-E1	AP-N1-FXS8 AP-N1-FXO8 AP-N1-FXS4O4	AP-N1-FXS8 AP-N1-FXO8 AP-N1- FXS4O4
Analog Ports	Up to 8	Up to 16	Up to 4	Up to 8	Up to 16	Up to 24
Signaling	SIP, H.323	SIP, H.323	SIP, H.323	SIP, H.323	SIP, H.323	SIP, H.323
Digital E1/T1	Up to 2E1	Up to 2E1	N/A	Up to 2E1	N/A	N/A
E&M	Support	N/A*	Support	Support	Support	N/A*
Module Slot	Two(2)	Two(2)	One(1)	Two(2)	Two(2)	Three(3)
LAN Port	2	2	2	2	2	2
Console	1	1	1	1	1	1
Power	Single PSU	Single PSU	Single PSU	Single PSU	Single PSU	Single PSU

VoIP Gateway (~32Port Analog, 2E1/T1)

Product	AP2340 	AP2640 	AP2650 
Available Modules	AP-N1-FXS8 AP-N1-FXO8 AP-N1-FXS4O4	AP-FXS8 AP-FXO8 AP-FXS4O4 AP-E&M8 AP-E1	AP-FXS8 AP-FXO8 AP-FXS4O4 AP-E&M8 AP-E1
Analog Ports	Up to 32	Up to 32	Up to 32
Signaling	SIP, H.323	SIP, H.323	SIP, H.323
Digital E1/T1	N/A	Up to 2E1	Up to 2E1
E&M	N/A*	Support	Support
Module Slot	Four(4)	Four(4)	Four(4)
LAN Port	2	2	2
Console	1	1	1
Power	Single PSU	Single PSU	Dual PSU

Large Capacity VoIP Gateway (~256Port)

Product	AP3100P 	AP6500 	AP6800 
Available VoIP Modules	AP-FXS4, AP-FXO4 AP-FXS2O2, AP-E&M4	AP-N1-FXS32 AP-N1-FXO32	AP-N1-FXS32 AP-N1-FXO32
Analog Ports	Up to 60 (4Port Module x 15)	Up to 128 (32 Port Module x 4)	Up to 256 (32 Port Module x 8)
Signaling	SIP, H.323	SIP, H.323	SIP, H.323
CPU Redundancy (Dual CPU)	N/A	Support (Option)	Support (Option)
E&M	Support	N/A	N/A
Module Slot for VoIP Module	15 Slots	4 Slots	8 Slots
LAN Port	2	2	2
Console	1	1	1
Dual Power Supply (Option)	Support	Support	Support

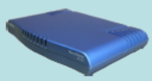
Digital VoIP Gateway (1E1/T1 ~ 16E1/T1 Port)

Product	AP1850 	AP-MG3000 	AP-MG3800 	AP-MG5000 
Available Modules	AP-N1-E1 AP-N1-2E1	APv2-1E1 APv2-2E1 APv2-4E1	HIM-VoIP4E1 (4 E1/T1 Module)	HIM-4E1 (4 E1/T1 Module)
VoIP Signaling	SIP, H.323	SIP, H.323	SIP, H.323	SIP, H.323
Digital E1/T1	Up to 4E1	Up to 4E1	Up to 8 E1	Up to 16E1
Digital Signaling	ISDN PRI, R2	ISDN PRI, R2	ISDN PRI, R2	ISDN PRI, R2
Module Slot	Two(2)	Two(2)	Two(2)	Four(4)
LAN Port	2	2	2	2
Console	1	1	1	1
Power	Single PSU	Single PSU	Single PSU	Dual PSU

SOHO VoIP Gateway Solution

Various SOHO VoIP gateway series of AddPac is approved for its high performance and reliability in world-wide markets. AddPac provides highly advanced VoIP services in order to meet demanding and evolving business requirements of customers. AddPac SOHO VoIP Gateway maintains best voice quality under not only high network bandwidth but also low network bandwidth using the latest voice compression algorithm and enhanced QoS (Quality of Service) management features of AddPac Technologies. Especially AddPac VoIP Gateway has been developed through years of experiences and accumulated know-how from existing enterprises and communication markets. AddPac VoIP Gateway provides concurrent dual stack such as H323, SIP and various voice codec such as G.711, G.729, G.723,etc for high quality of communication service. This will satisfy customers with high expectation.

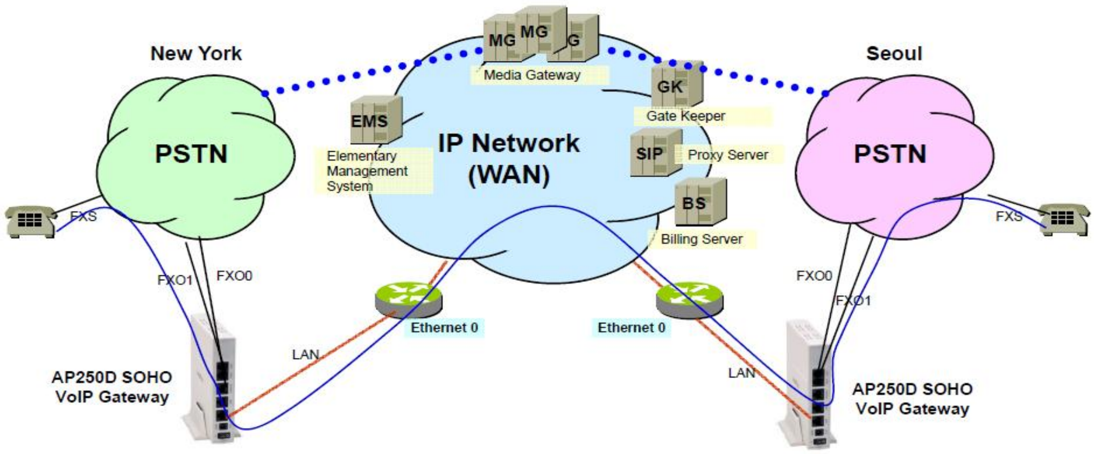
VoIP Gateway for SOHO (1~2Port)

Product	AP100 		AP100B 		AP190 	AP200 		AP250 	
Model	Type	VoIP	Type	VoIP		Type	VoIP	Type	VoIP
	P	PSTN Backup	B	2FXS, PSTN Backup		B	2FXS, PSTN Backup	B	2FXS, PSTN Backup
						D	2FXO	D	2FXO
						E	1FXS, 1FXO		
VoIP Ports	1		2		1	2		2	
Signaling	SIP, H.323		SIP, H.323		SIP,H.323	SIP, H.323		SIP, H.323	
Module Slot	N/A		N/A		N/A	N/A		N/A	
LAN Port	2		2		2	2		2	
Console	N/A		N/A		Support	Support		Support	
Power	External Adaptor		External Adaptor		External Adaptor	External Adaptor		External Adaptor	

VoIP Gateway for SMB (4Port)

Product	AP700 		AP1000 	AP1002 	AP1005 	AP800 	
Model	Type	VoIP				Type	VoIP
	P	1-Port PSTN Backup				A	1-Port PSTN Backup
						B	4-Port PSTN Backup
VoIP Ports	4-Port FXS		4-Port FXS	2-Port FXS & 2-Port FXO	4-Port FXO	4-Port FXS	
Signaling	SIP, H.323		SIP, H.323	SIP,H.323	SIP, H.323	SIP, H.323	
Module Slot	N/A		N/A	N/A	N/A	N/A	
LAN Port	2		2	2	2	2	
Console	Support		Support	Support	Support	Support	
Power	External Adaptor		External Adaptor	External Adaptor	External Adaptor	External Adaptor	

Network Diagram



E&M VoIP Gateway Solution

Various E&M VoIP gateway series of AddPac is approved for its high performance and reliability in world-wide markets. AddPac provides highly advanced VoIP services in order to meet demanding and evolving business requirements of customers. Especially AddPac E&M VoIP Gateway has been developed through more than 10 years of experiences and accumulated know-how from existing enterprises and communication markets. AddPac E&M VoIP Gateway provides concurrent triple stack such as H323, SIP, MGCP and QoS (Quality of Service) for high quality of communication service. This will satisfy customers with high expectation.

E&M VoIP Gateway (~24Port Analog)

Product	AP1700 	AP2610 	AP2620 	AP2120N 
Available Modules	AP-FXS4 AP-FXO4 AP-FXS2O2 AP-E&M4 AP-E1	AP-FXS4 AP-FXO4 AP-FXS2O2 AP-E&M4	AP-FXS4 AP-FXO4 AP-FXS2O2 AP-E&M4 AP-E1	AP-N1-FXS8 AP-N1-FXO8 AP-N1-FXS4O4 AP-E&M8
Analog Ports	Up to 8	Up to 4	Up to 8	Up to 16
Signaling	SIP, H.323	SIP, H.323	SIP, H.323	SIP, H.323
Digital E1/T1	Up to 2E1	N/A	Up to 2E1	N/A
E&M	Support	Support	Support	Support
Module Slot	Two(2)	One(1)	Two(2)	Two(2)
LAN Port	2	2	2	2
Console	1	1	1	1
Power	Single PSU	Single PSU	Single PSU	Single PSU

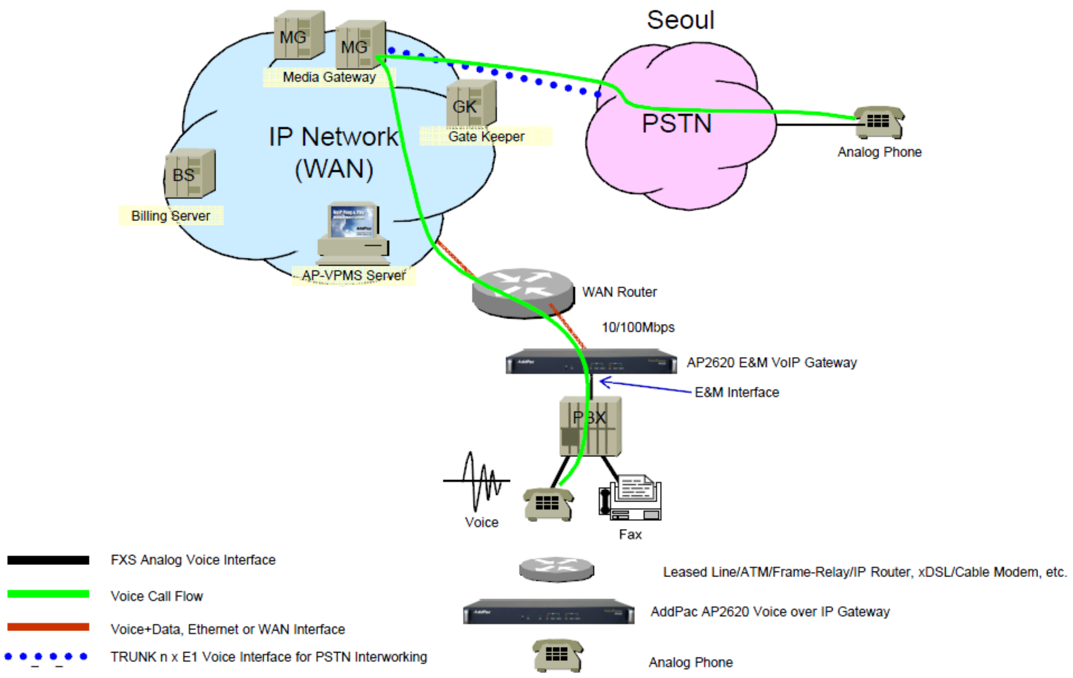
E&M VoIP Gateway (~32Port Analog)

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

E&M VoIP Gateway (~60Port)

Product	AP3100P 
Available VoIP Modules	AP-FXS4, AP-FXO4 AP-FXS2O2, AP-E&M4
Analog Ports	Up to 60 (4Port Module x 15)
Signaling	SIP, H.323
CPU Redundancy (Dual CPU)	N/A
E&M	Support
Module Slot for VoIP Module	15 Slots
LAN Port	2
Console	1
Dual Power Supply (Option)	Support

Network Diagram



Digital VoIP Gateway Solution

Various Digital VoIP gateway series of AddPac is approved for its high performance and reliability in world-wide markets. AddPac provides highly advanced VoIP services in order to meet demanding and evolving business requirements of customers. Especially AddPac Digital VoIP Gateway has been developed through more than 10 years of experiences and accumulated know-how from existing enterprises and communication markets. AddPac Digital VoIP Gateway provides concurrent triple stack VoIP Signaling such as H323, SIP, MGCP, and supports ISDN-PRI, R2, SS7* Digital Interface Signaling. Also AddPac Digital VoIP Gateway supports QoS (Quality of Service) for high quality of VoIP communication service. This will satisfy customers with high expectation.

Digital VoIP Gateway (1~2Port)

Product	AP1700 	AP2620 	AP2640 	AP2650 
Available Modules	AP-FXS4 AP-FXO4 AP-FXS2O2 AP-E&M4 AP-E1 AP-T1	AP-FXS4 AP-FXO4 AP-FXS2O2 AP-E&M4 AP-E1 AP-T1	AP-FXS8 AP-FXO8 AP-FXS4O4 AP-E&M8 AP-E1 AP-T1	AP-FXS8 AP-FXO8 AP-FXS4O4 AP-E&M8 AP-E1 AP-T1
Analog Ports	Up to 8	Up to 8	Up to 32	Up to 32
VoIP Signaling	SIP, H.323	SIP, H.323	SIP, H.323	SIP, H.323
Digital E1/T1	Up to 2E1	Up to 2E1	Up to 2E1	Up to 2E1
Digital Interface Signaling	ISDN PRI, R2	ISDN PRI, R2	ISDN PRI, R2	ISDN PRI, R2
Module Slot	Two(2)	Two(2)	Four(4)	Four(4)
LAN Port	2	2	2	2
Console	1	1	1	1
Power	Single PSU	Single PSU	Single PSU	Dual PSU

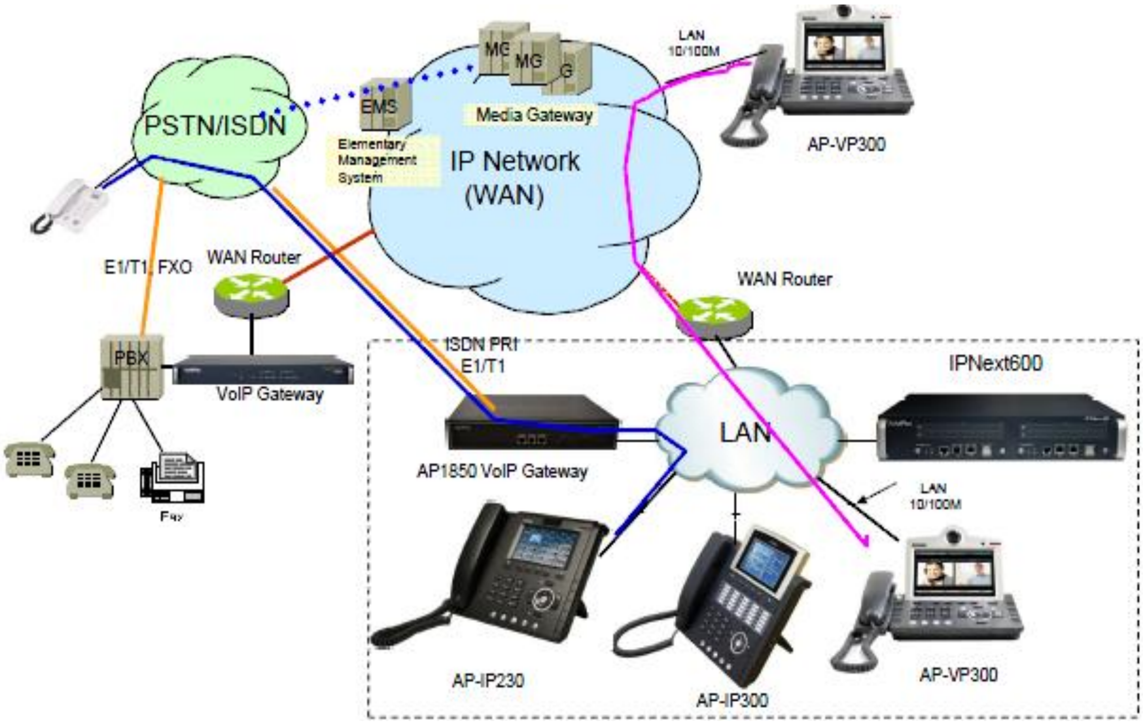
Digital VoIP Gateway (1~4Port)

Product	AP1800 	AP1850 
Available Modules	AP-N1-FXS8 AP-N1-FXO8 AP-N1-FXS4O4 AP-N1-E1, AP-N1-2T1 AP-N1-2E1, AP-N1-2T1	AP-N1-E1, AP-N1-2T1 AP-N1-2E1, AP-N1-2T1
Analog Ports	Up to 16	N/A
VoIP Signaling	SIP, H.323	SIP, H.323
Digital E1/T1	Up to 2E1	Up to 4E1
Digital Interface Signaling	ISDN PRI, R2, SS7*(Option)	ISDN PRI, R2, SS7*(Option)
Module Slot	Two(2)	Two(2)
LAN Port	2	2
Console	1	1
Power	Single PSU	Single PSU

Digital VoIP Gateway (1~16Port)

Product	AP-MG3000 	AP-MG3800 	AP-MG5000 
Available Modules	APv2-1E1 APv2-2E1 APv2-4E1	HIM-VoIP4E1 (4 E1/T1 Module)	HIM-4E1 (4 E1/T1 Module)
VoIP Signaling	SIP, H.323	SIP, H.323	SIP, H.323
Digital E1/T1	Up to 4E1	Up to 8 E1	Up to 16E1
Digital Signaling	ISDN PRI, R2	ISDN PRI, R2	ISDN PRI, R2
Module Slot	Two(2)	Two(2)	Four(4)
LAN Port	2	2	2
Console	1	1	1
Power	Single PSU	Single PSU	Dual PSU

Network Diagram



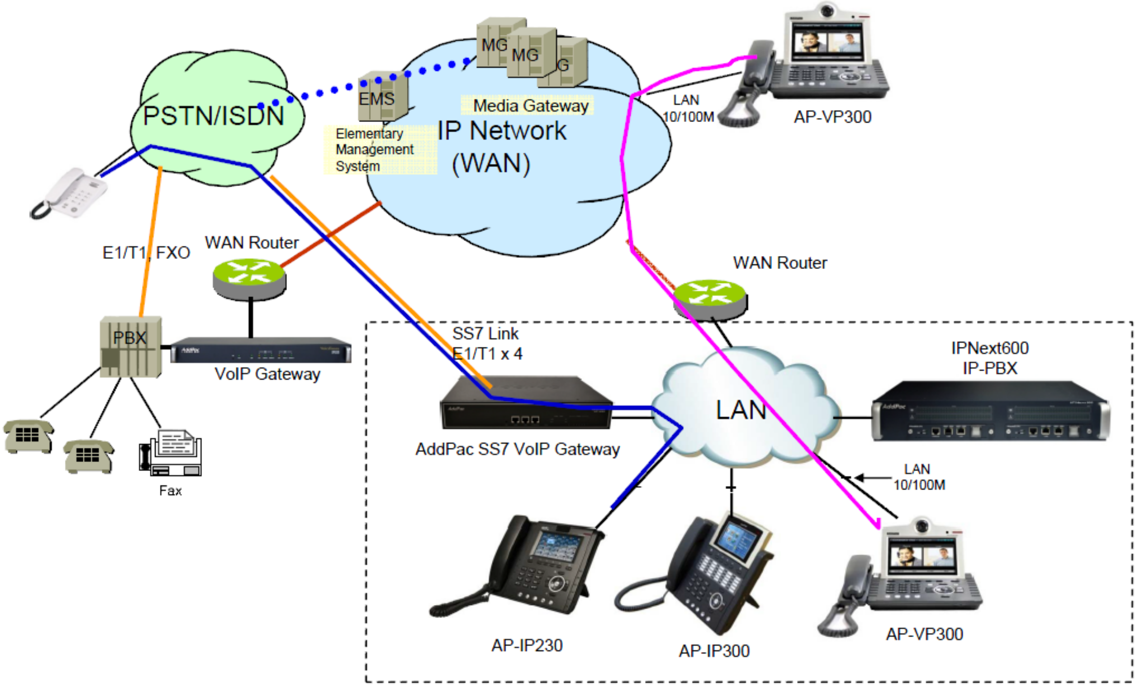
SS7 VoIP Gateway Solution

Various SS7(No.7) digital VoIP gateway series of AddPac is approved for its high performance and reliability in world-wide markets. AddPac provides highly advanced VoIP services in order to meet demanding and evolving business requirements of customers. Especially AddPac SS7 digital VoIP Gateway has been developed through more than 10 years of experiences and accumulated know-how from existing enterprises and communication markets. AddPac SS7 VoIP Gateway provides concurrent triple stack VoIP Signaling such as H323, SIP, MGCP, and supports ISDN-PRI, R2, SS7 Digital Interface Signaling. Also AddPac SS7 VoIP Gateway supports QoS (Quality of Service) for high quality of VoIP communication service. This will satisfy customers with high expectation.

Digital VoIP Gateway (1~16Port)

Product	AP1800 	AP1850 	AP-MG3000 	AP-MG3800 	AP-MG5000 
Available Modules	AP-N1-FXS8 AP-N1-FXO8 AP-N1-FXS4O4 AP-N1-E1	AP-N1-E1 AP-N1-2E1	APv2-1E1 APv2-2E1 APv2-4E1	HIM-VoIP4E1 (4 E1/T1 Module)	HIM-4E1 (4 E1/T1 Module)
VoIP Signaling	SIP, H.323	SIP, H.323	SIP, H.323	SIP, H.323	SIP, H.323
Digital E1/T1	Up to 2E1	Up to 4E1	Up to 4E1	Up to 8 E1	Up to 16E1
Digital Signaling	ISDN PRI, R2, SS7	ISDN PRI, R2, SS7	ISDN PRI, R2, SS7	ISDN PRI, R2, SS7	ISDN PRI, R2, SS7
Module Slot	Two(2)	Two(2)	Two(2)	Two(2)	Four(4)
LAN Port	2	2	2	2	2
Console	1	1	1	1	1
Power	Single PSU	Single PSU	Single PSU	Single PSU	Dual PSU

Network Diagram



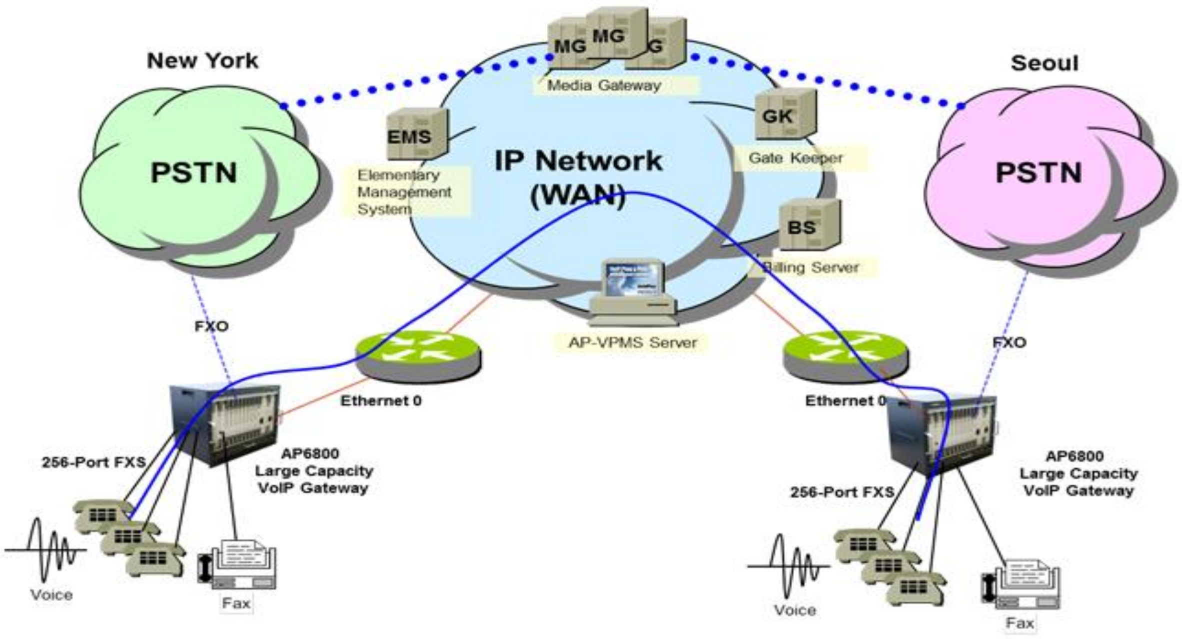
Large Capacity VoIP Gateway Solution

The performance and reliability of AddPac Large Capacity VoIP gateway series have been recognized in global markets as well as domestic market. Large Capacity VoIP Gateway such as AP6800, which is a collection of experiences and know-how accumulated in the enterprise and service provider markets, provides excellent services that meet the needs of customers that ask for high quality VoIP services. Also, this product provides better VoIP services due to the unique QoS algorithm of AddPac by ensuring Quality of Service (QoS) for SIP, MGCP VoIP call control protocols, and better calling quality. The analog interfaces of AddPac Large Capacity VoIP Gateway present an optimal call scenario for inter-working with an analog Legacy PBX and Call Manager for IP telephony, and ensure better features and performance compared to the plain old VoIP gateways.

Large Capacity VoIP Gateway (~256Port)

Product	AP3100P 	AP6200 	AP6500 	AP6800 
Available VoIP Modules	AP-FXS4, AP-FXO4 AP-FXS2O2, AP-E&M4	AP-N1-FXS8, AP-N1-FXO8, AP-N1-E&M4, AP-N1-FXS4O4, AP-N1-E1/T1, AP-N1-2E1/T1	AP-N1-FXS32 AP-N1-FXO32	AP-N1-FXS32 AP-N1-FXO32
Analog Ports	Up to 60 (4-Port Module x 15)	Up to 80 (8-Port Module x 10)	Up to 128 (32-Port Module x 4)	Up to 256 (32-Port Module x 8)
Signaling	SIP, H.323	SIP, H.323	SIP, H.323	SIP, H.323
CPU Redundancy (Dual CPU)	N/A	N/A	Support (Option)	Support (Option)
E&M	Support	Support	N/A	N/A
Module Slot for VoIP Module	15 Slots	10 Slots	4 Slots	8 Slots
LAN Port	2	2	2	2
Console	1	1	1	1
Dual Power Supply (Option)	Support	Support	Support	Support

Network Diagram



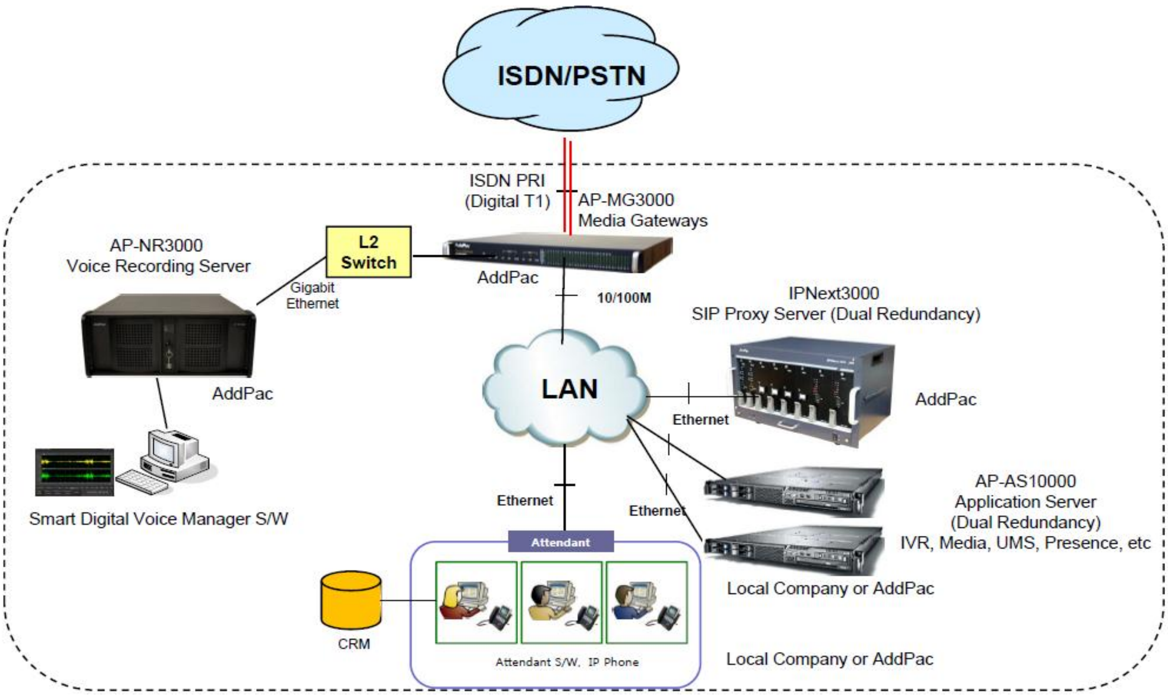
Media Gateway Solution

The performance and reliability of AddPac Media gateway series have been recognized in global markets as well as domestic market. Media Gateway such as AP-MG5000, which is a collection of experiences and know-how accumulated in the enterprise and service provider markets, provides excellent services that meet the needs of customers that ask for high quality VoIP services. Also, this product provides better VoIP services due to the unique QoS algorithm of AddPac by ensuring Quality of Service (QoS) for SIP, MGCP VoIP call control protocols, and better calling quality. The digital interface of AddPac Media Gateway present an optimal call scenario for inter-working with an Legacy PBX and Call Manager for IP telephony, and ensure better features and performance compared to the plain old VoIP gateways.

Media Gateway (1~16 E1/T1)

Product	AP-MG3000 	AP-MG3800 	AP-MG5000 
Available Modules	APv2-1E1 APv2-2E1 APv2-4E1	HIM-VoIP4E1 (4 E1/T1 Module)	HIM-4E1 (4 E1/T1 Module)
VoIP Signaling	SIP, H.323	SIP, H.323	SIP, H.323
Digital E1/T1	Up to 4E1	Up to 8 E1	Up to 16E1
Digital Signaling	ISDN PRI, R2	ISDN PRI, R2	ISDN PRI, R2
Module Slot	Two(2)	Two(2)	Four(4)
LAN Port	2	2	2
Console	1	1	1
Power	Single PSU	Single PSU	Dual PSU

Network Diagram



Secure VoIP Gateway Solution

Various VoIP gateway series of AddPac is approved for its high performance and reliability in world-wide markets. AddPac provides highly advanced VoIP services in order to meet demanding and evolving business requirements of customers. Especially AddPac Secure VoIP Gateway has been developed through years of experiences and accumulated know-how from existing enterprises and communication markets. AddPac Secure VoIP Gateway provides concurrent triple stack such as H323, SIP, MGCP and QoS (Quality of Service) for high quality of communication service. This will satisfy customers with high expectation. VoiceFinder secure VoIP gateway support analog and digital (T1,E1) voice interface. AddPac secure VoIP gateway supports optimum level of call scenario when interworking with Legacy PBX. Also, it is 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.

Secure Analog VoIP Gateway (Up to 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

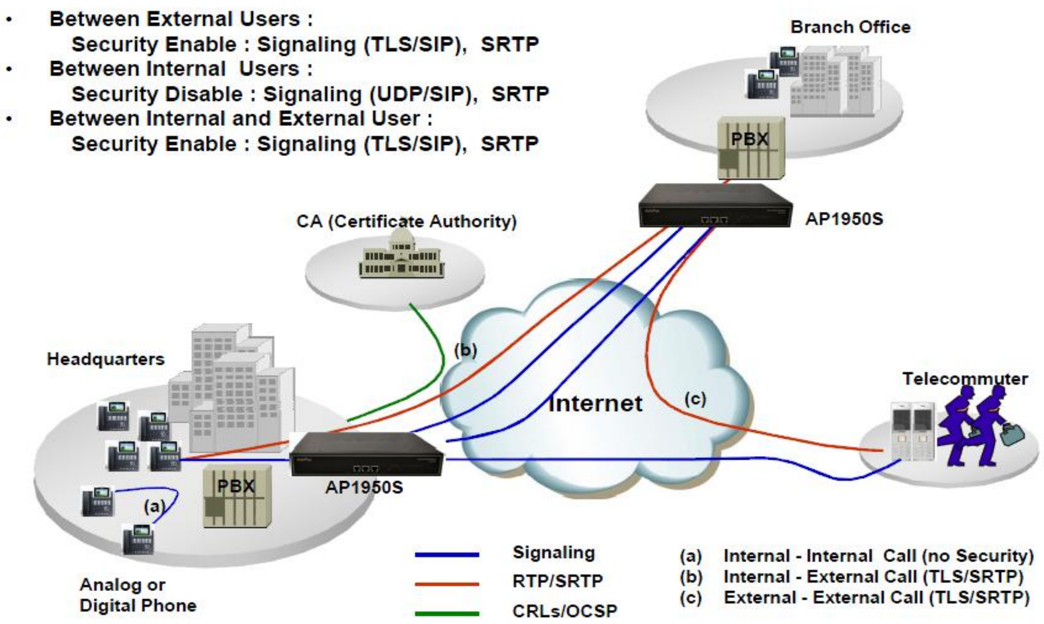
Secure Digital/Analog VoIP Gateway (Up to 2E1)

Product	AP1900S 	AP1950S 
Available Modules	AP-N1-E1 AP-N1-FXS8 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

Comparison between Secure VoIP Gateway and VPN solution

Category	VPN + VoIP Gateway Security	Secure VoIP Gateway Security
Encrypted Data	Entire Network Data (Possible Delay)	Only VoIP Voice Data
Secured Area	Apply in general	Only selected areas, (Ordinary VoIP calls are possible for other areas)
GateKeeper Requirement	Internal GK is mandatory	Not essential Able to utilize external public GK
Origin of the encryption process	VPN Device	Secure VoIP Gateway
Center Line Error	Failure on the entire network	Failure on calls connected to main office
Network configuration	Low (In a large-scale VoIP Network, it cannot emerged into mesh structure.)	High (Can be easily formed as mesh type and be expanded easily)
Investment	High (VPN device and additional GK requires)	LOW

Network Diagram



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