High Performance Audio MCU Solution

AP-MC1500

AP-MC1500 High Performance Audio MCU

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, IPNext5000, etc), IP terminal equipments such as AP-IP300, AP-IP160, AP-IP120 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 suit 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. AP-MC1500 is designed on the basis of parallel DSP architecture with the highest level processing capabilities for large-capacity multipoint audio conferencing. HIM-AMCU128 audio MCU module can support maximum 128 audio channels (IP end-point terminal) concurrently, and HIM-AMCU64 module can support maximum 64 audio channels concurrently. AP-MC1500 supports diverse audio codecs concurrently which is ordinarily being used on IP end-point terminals. It supports mixed IP audio calls to enable the powerful mixing/transcoding capabilities among diverse audio codecs simultaneously providing the optimal audio experience. AP-MC1500 supports the multiple audio conferencing sessions and multiple audio codecs (G.711, G.726, G.729, G.723.1,etc) per a session. Also AP-MC1500 supports the SIP, H.323 protocols for VoIP signaling. For H.323 based VoIP signaling, AP-MC1500 supports the internal H.323 gatekeeper service features. For SIP VoIP signaling protocols, external AddPac Technology's enterprise SIP Call Manager is recommended.

AP-MC1500 is an integrated, feature-rich network equipment delivering routing, NAT/PAT, DHCP Server/Relay, Public IP sharing, VRRP and QoS. In today's mixed network of xDSL, Cable, FTTH, Metro Ethernet, Metro ATM, Leased line and dynamic IP environment, not only the ample network service features, but also high-end QoS(Quality of Service) and security features are requested. Based on two(2) 10/100Mbps Fast Ethernet ports, AP-MC1500 offers integrated network and security service of LAN-to-LAN routing, bridge and NAT/PAT.

Main Features

- AP-MC1500 is a audio conference MCU equipment that allows you to build a multimedia system in order to integrate voice, audio, and data.
- Up to 128/256 Party Audio Conference Support
- Two(2) module slots support for audio conference modules, network interface modules.
- Audio MCU, Network Interface Modules
  -HIM-AMCU128
  :128 channel voice MCU module
  -HIM-AMCU64
  : 64 channel voice MCU module
  - HIM-ATMOC3
  :1-Port OC3 155Mbps ATM Interface Module
  - HIM-ATMDS3
  : 1-Port DS3 45Mbps ATM Interface Module
  - HIM-V35FR2
  : 2-Port V.35 Interface Module
  - HIM-V35FR6
  : 6-Port V.35 Interface Module
- Supports the state-of-art codec algorithm for voice and audio services such as G.711, G.726, G.729, G.723.1,etc.
- Dynamic Session Management Support
- Meet-Me, Add-Hoc, Dial-Out Video Conferencing Signaling Support.
- SIP, H.323 VoIP Signaling Support
- Ensures the best voice quality on a regular IP-based Internet network that provides an asymmetrical bandwidth.
- Two(2) 10/100Mbps Fast Ethernet Interface Support
- Allows you to set up configuration and your network environment by MS-Window based Smart Multimedia Manager S/W
- Ensures extensibility, reliability, and authenticity since the APOS inter-networking software of AddPac is installed.

AP-MC1500 Applications

- IP audio conferencing system
- IP Telephony System

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General Hardware Specifications

CPU	High Performance RISC Integrated Host Processor
Memory	Flash : 4/8Mbyte SDRAM : 64MByte
Network Interface	Two(2)-port 10/100Mbps Fast Ethernet (LAN0, LAN1)
Console Interface	1-port RS-232C Console Interface
Operation Environment	Temperature 0°C ~+45°C (operating), -40°C ~+85°C (storage), Humidity 5% ~ 95%
Power Supply	AC110~220VAC 50/60Hz, 5V 8A Free Voltage
Dimension	66mm x 482mm x 390mm (H x W x D)
Weight	6.48Kg

Audio MCU Service Features

MCU Mode	Local MCU Mode, Remote MCU Mode
Local MCU Mode	Standalone MCU Mode Support, Internal Gatekeeper H.323 Signaling Support, SMM (Smart Multimedia Management) program Support for MCU Management
Remote MCU Mode	SIP Server (or H.323 Gatekeeper) + Audio MCU AddPac SIP Server, GateKeeper Interworking Mode Audio Mixing SIP Server : SIP Signaling Gatekeeper : H.323 Signaling
Audio Conferencing Service	Add-Hoc, Meet-Me, Dial-out Conferencing Mode Support

Audio MCU Module Features

HIM-AMCU128 128-channel Audio MCU Module

High quality Audio Mixing
128 Channel Audio Mixing
Compact PCI Style Hot-Swap Function
High-End Programmable DSPs
Parallel DSP Processing for High Quality Audio Mixing
Concurrent Different Audio Codec Support

HIM-AMCU64 64-channel Audio MCU Module

High quality Audio Mixing
64 Channel Audio Mixing
Compact PCI Style Hot-Swap Function
High-End Programmable DSPs
Parallel DSP Processing for High Quality Audio Mixing
Concurrent Different Audio Codec Support

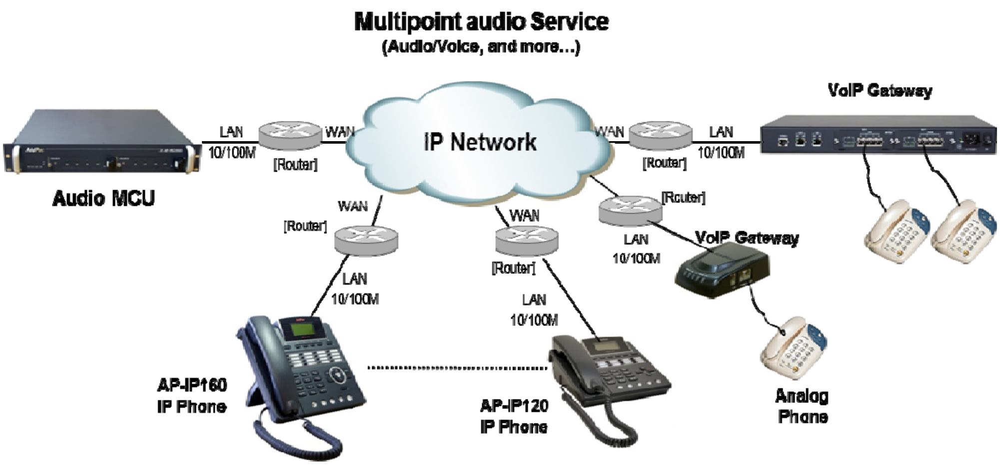
Audio Service and Signaling Protocol Features

Voice and Audio Codec	G.711, G.723.1, G.726, G.729 ,etc
VoIP Signaling Protocol	SIP/H.323 VoIP Signaling Protocol (Dual Stack) ITU-T H.323 Gateway, Gatekeeper Support SIP Proxy Server Interoperability
Voice Processing	VAD, CNG, Dynamic Jitter Buffer Operation
DTMF	Detection and Generation, RFC 2833 Compliant
Voice QoS	Enhanced QoS Management for Voice Traffic

WAN, LAN, IP Services and Other Features

WAN Protocol	Point-to-Point Protocol (PPPoE) for ADSL
IP Routing	IPv4 Static and IEEE 802.1Q VLAN Routing, RIP, OSPFv2, etc
Network Management	Standard SNMP Agent (MIB v2) Support, Console, Telnet, Remote Firmware Upgrade via FTP/TFTP Support
Security Features	IP Packet Filtering, Access List, Access Control and Data Protections, Enable/Disable for Specific-Protocols Multi-level User Account Management
Operation & Management	Configuration Backup and Restore Management, Debugging and Diagnosis Features, System Booting/Rebooting through Watch-Dog, etc.
Other Features	DHCP Server and Relay, Network Address Translation (NAT), Port Address Translation (PAT), IEEE Standard Transparent Bridging, CLI, etc.

Network Diagram



Ordering Information

- AP-MC1500 High Performance Audio MCU with optional two(2) Module slots

- CAB-LAN Ethernet Cable
- CAB-CON RS-232C Console Cable
- SMM S/W Smart Multimedia Manager S/W
- HIM-AMCU128 128 ch. audio MCU module
- HIM-AMCU64 64 ch. audio MCU module
- HIM-ATMOC3 1-Port ATM OC3 module
- HIM-ATMDS3 1-Port ATM DS3 module
- HIM-V35FR2 2-Port V35 Interface module
- HIM-V35FR6 6-Port V35 Interface module

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