Product Name: Xorcom Video Conferencing and PBX system - 2 x PRI (T1 or E1) port + HWEC XV3055
Manufacturer: Xorcom
Model Number: XV3055

Please Note: This Xorcom XV3000 has been discontinued. For an alternative, we recommend the IP Video Conferencing range of products.

Xorcom Video Conferencing and PBX system - 2 x PRI (T1 or E1) port + HWEC XV3055
The Xorcom Video Conferencing and PBX system - 2 x PRI (T1 or E1) port + HWEC XV3055 enables organizations to conduct high quality multi-point video conferencing at an affordable price. The solution has been integrated seamlessly into Xorcom’s award-winning Asterisk based business telephony solution. SIP (Session Initiation Protocol) signaling ensures compatibility with a wide variety of endpoints, including the leading IP-based video phones as well as soft phones, significantly reducing the total cost of the solution.
Key Features of Xorcom Video Conferencing and PBX system - 2 x PRI (T1 or E1) port + HWEC XV3055

• High Definition (HD) Voice and Video - Supports resolutions of up to 720p and Wideband HD Voice for unmatched voice quality bridge and density.
• Unmatched Price and Value - Organizations that do not have video and voice conferencing solutions cannot afford not to have it!
• Simple Solution - Built to be installed and up and running within minutes in any IP network environment, using simple Web management
• Seamless Interoperability - Built on a field-hardened media processing foundation; provides standard-based SIP protocol to connect different devices
• Smooth Transcoding - Transcoding and transrating between different endpoints with unmatched media quality
• Intuitive Web Management - Provides full control to all levels of users, from IT managers to any user in the organization
• Reliable, Flexible Telephony - Unique PBX design ensures the reliability of a proprietary telephony system, while providing the flexibility, scalability, easy integration and competitive pricing afforded by an open source platform

 XV3055 MCU Product Specification Highlights
Standards-based Signaling/Coding

• Able to connect different devices, from room systems to desktop video phones, to video clients running on smart phones or tablets. Transcodes the video conference providing the best available video conference quality to every participant.

Signaling

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SIP (Session Initiation Protocol), according to the following RFCs: RFC3261, RFC4566, RFC2976, RFC3262, RFC3263, RFC3264, RFC4317, RFC3581, RFC3966, RFC4028, RFC2833, RFC3550, RFC3551, RFC 3951, RFC3952

Conferencing Types

• Support for both voice-only bridge or video conferencing capabilities

Voice Codecs

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G711a, G711u, G729, G722.1, G723, iLBC

Video Codecs

• MPEG4, H263, H264

Video Resolution

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QCIF, CIF, VGA, 4CIF and HD (720p)

Great Performance for SMB*

• Normal Conference Room (CIF) 1 conference room instance with 15 participants/12 viewable, or 2 simultaneous conference rooms with 15 participants/6 viewable
• High Conference Room (VGA) 1 conference room instance with 15 participants/10 viewable
• HD Conference Room (720p) 1 conference room instance with 15 participants/5 viewable
• Voice-only Conference Up to 12 simultaneous conference rooms with 30 participants per room

Transmitted Bandwidth from MCU

• HD Participant: 2 Mbps HD (1280X720) 15 FPS
• VGA Participant: 1.2 Mbps VGA (640X480) 30 FPS (High Conference) / 15 FPS (HD Conference)
• CIF Participant: 320Kbps CIF (352X288) 24 FPS PM0016.01

Minimum Bandwidth to MCU
The minimum required bandwidth from connecting endpoints is as follows:

• HD Participant: 1.2 Mbps
• VGA Participant: 320 Kbps
• CIF Participant: 320 Kbps

Configurable Meeting Rooms

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A conference meeting room can be provisioned at any time, and used at any time. The conference rooms can be configured as voiceonly or voice and video conference rooms.

Multiple Access Numbers

• Supports multiple access numbers, which can be registered in a SIP network as an endpoint or SIP trunk.

VIP Participants

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Participants may be recognized by the system and automatically connected to a predefined bridge; useful when connecting multiple conference rooms to a preset configuration.

Leader-based Moderation
Conference rooms can be moderated by a leader or set as unmanaged, to be used as needed. Dashboard enables the moderator to do the following:

• Set the dominant speaker
• View participant media information
• Mute/Unmute Voice
• Mute/Unmute Video
• Disconnect a participant
• Mute/Unmute all participants
• Invite multiple participants (Dial Out)
• Change the default layout to Equal or Dominant-based layout

Dynamic Video Layouts

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As video participants join/leave the conference, the layouts automatically adjust to provide the optimal coverage of all viewable participants.

Energetic Voice Detection – Dominant-Based Layout

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Participant placement within the dominant speaker layout is updated based on continual measurement of all active voice participants. The resulting top participants are displayed.

VIP Participants

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Participants may be recognized by the system and automatically connected to a predefined bridge; useful when connecting multiple conference rooms to a preset configuration.

Please Enquire