

Macrotel x1 x2 Multimode



POTS, VOIP, Bluetooth or GSM professional HD Audio hybrid

The new generation of Macrotel X introduces the AxelTech Multimode Technology. The internal framework based on DSP (Digital Signal Processor) allows to work in real time achieving the highest audio performance.

This Technology sets Macrotel X1 Multimode and Macrotel X2

Multimode as cutting-edge equipment in the telephone interface technology and allows managing one or two telephone connections: POTS (Plain Old Telephone Service) landline, VOIP, Bluetooth or GSM (with optional interface). With Macrotel X2 Multimode any technology combination is possible.

Audio enhancement features are custom designed to guarantee the best quality in phone calls. Thanks to AxelTech's ten-year experience in the field of audio processing we should consider this product not a common telephone hybrid but a Telephone Audio Processor able to shape the sound to optimize the yield of every single phone call. In fact the new Macrotel X Multimode series integrate a powerful DSP system that takes care of the telephone signal process:

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- Automatic Gain control (AGC)
- Echo canceller
- Hold caller/Attenuator
- Expander and Compressor
- Audio Limiter
- 3 Bands Fully parametric **Equalizer**

The new Macrotel X Multimode range does not require the installation of any software: the unit has an integrated Web server supporting the most popular browsers (Chrome, Firefox, Explorer, Opera, etc.).

A simple and intuitive GUI can be accessed through any device: PC, notebook, tablet, smartphone, etc.

For services synchronization and logs the Macrotel connects to a NTP server.

The main XLR input and output connectors can manage analogue or AES/EBU signals according to the device settings (AES/EBU input/output is optional).

A second XLR analogue output is anyway available. In Bluetooth mode the two outputs act like a stereo balanced output.

Macrotel X Multimode is able to record the telephone calls

audio on a USB flash memory plugged in the USB front panel

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slot.

Audio will be recorded in PCM format with a self-explaining filename based on date/time and can include only the RX audio or both RX and TX signals.

Moreover it's possible to send an RTP-PCM audio stream to an external PC to record the call.

This recording can include only the RX audio or both RX and TX signals.

A multiple GPIO port allows to manage the device from external equipment, like audio console or PC. System sends "Ring", "Hold", "Hook" and "REC" status and manages "Hook", "Rec", "Mode" and "Hold" functions.

On the front panel an audio meters with 18 LEDs shows the RX and TX audio level of each channel. Additional LEDs show working Mode and Power on.

Power supply can be between 90 and 260 V AC - 50/60 Hz: this allows to use it worldwide.

Models





Macrotel X1 Multimode

MACROTEL X1 PSTNVOIP/8T	







Macrotel X2 Multimode

MACROT PSTN/VOIP/BT 18 12 4	EL X2 TX MODE 0 +6 -38 -32 -6 0 +6 P015 BT W	RX TX HOOK		TX MODE RX -18 -12 -6 0 +6 POTS BT VOIP	TK C 2
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Macrotel X1 & X2 Multimode

Main Features

Analog and AES/EBU digital inputs and outputs
1 line and 2 lines models
POTS/PSTN and GSM Quad Band (option)
Integrated web server for remote control
Digital AGC processor with 3 bands fully parametric EQ
Digital Echo canceller
Separate send and receive
LAN and USB ports
Auto answer and disconnection
Balanced XLR I/O (mic/line)
Remote control software and dialer

Highlights

Multi-Line digital telephone hybrid POTS/GSM/VOIP Echo cancellation through a digital process on DSP (POTS) Advanced audio processing functions: AGC, Parametric EQ, Audio filters, Compressor, Expander add Limiter Internal Web Server allows device configuration Automatic setup



1 selectable XLR input: Analog or AES/EBU (AES/EBU is optional) 1 selectable XLR output: Analog or AES/EBU (AES/EBU is optional) 1 balanced XLR analogue output Phone call recording in PCM format on USB support (RX only or RX+TX) Front panel Led Meters displaying RX/TX levels Audio stream generation in RTP/PCM format (RX only or RX+TX) Front panel LED: Gain RX, Gain TX, Mode "Hook" and "Hold line" lighted buttons 4 GPI interfaces, 4 GPO interfaces Integrated Caller Identifier (CID) – only for VOIP Input and output call logs G711-G722 VOIP audio codecs Local telephone output (POTS)

General Functions

The integrated web server allows configuring the Macrotel X parameters through a web interface. Web user-friendly GUI is available on any kind of device: PC, notebook, tablet, smartphone. Macrotel X supports many TCP/IP standards such as HTTP for web GUI and UDP for streaming. Connection to external NTP server (Internal RTC with buffer) for time synch. Ethernet/GPIO connection available.



Stainless steel enclosure. Front panel with 22 levels and status

LEDs.

Power supply: 90-260V AC 50/60 Hz - 10 W (green approach).

POTS Module

Standard RJ11 sockets for connecting the telephone line and

external telephone set.

Internal DIAL system via DTMF for direct call (no need of an

external telephone set).

Logs phone numbers of incoming and outgoing calls (date, call time

start, call time end).

Exporting logs available on USB key.

The external telephone set can be release when the telephone

hybrid hooks the line.

Automatic telephone line hooking (after a settable number of rings).

Automatic telephone line release based on the "dropped line" tone

(this feature can be disabled).

Adjustable AGC, Compressor/Limiter and Equalizer parameters.

Adjustable Telephone band (Low or Hi-Cut).

Voip Module

Standard network RJ45 connector.

SIP version 2.0 (RFC 3261).

Authentication methods: SIP/IAX Dynamic Registration (Register) |



SIP Static IP authentication.

Supported audio codecs: G.711u (PCM u) | G.711a (PCM a) | G.722

| G.722.1 24/32.

Automatic telephone line hooking (after a settable number of rings).

Automatic telephone line release based on the "dropped line" tone

(this feature can be disabled).

Logs phone numbers of incoming and outgoing calls (date, call time

start, call time end).

Exporting logs available on USB key.

Adjustable AGC, Compressor/Limiter and Equalizer parameters.

Bluetooth Module

Fully certified Bluetooth version 3.0 audio module, compatible with

Bluetooth version 2.1 + EDR, 1.2, and 1.1.

SIP version 2.0 (RFC 3261).

Embedded Bluetooth stack profiles: A2DP and HFP/HSP.

Supports iAP profile discovery for iPhone® and iPod® Bluetooth

accessories G.711u.

Dual-channels differential audio input and output for highest

quality audio.

Certifications: FCC, IC, CE.

Adjustable AGC, Compressor/Limiter and Equalizer parameters.

Front Panel



MACROTEL X1

Stainless steel mechanic 1 Rack Unit standard 19"

USB connector

18+4 LEDs RX/TX meters

Mode selection button

RX and TX gain buttons (+ and -)

Hook and Hold buttons



MACROTEL X2

Stainless steel mechanic 1 Rack Unit standard 19" Two USB Connectors 18+4 LEDs RX/TX meters for each channel Mode selection button for each channel RX and TX gain buttons (+ and -) for each channel Hook and Hold buttons for each channel

Rear Panel





MACROTEL X1

Universal power supply 90-240V AC - 50/60Hz - 10 Watt

Ethernet port LAN/WAN on RJ45 connector

GPIO port on DB 9

RJ11 Landline connector

RJ11 Telephone set connector

Analog/AES/EBU Output (AES/EBU is optional)

Analog Output

Analog/AES/EBU Input (AES/EBU is optional)



MACROTEL X2

Universal power supply 90-240V AC - 50/60Hz - 2x10 Watt

Two Ethernet ports LAN/WAN on RJ45 connectors

Two GPIO ports on DB 9

Two RJ11 Landline connectors

RJ11 Telephone set connectors

Channel 1 Analog/AES/EBU Output | Channel 2 Analog/AES/EBU

Output

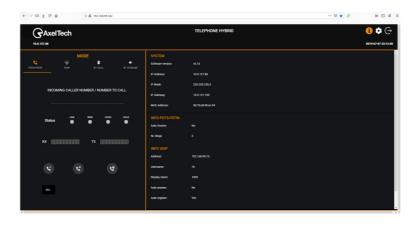
Channel 1 Analog Output | Channel 2 Analog Output



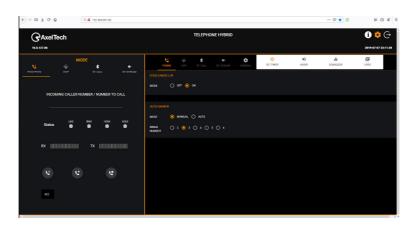
Channel 1 Analog/AES/EBU Input | Channel 2 Analog/AES/EBU

Input

Web Interface



Web Interface - Main Page



Web Interface - Phone Settings

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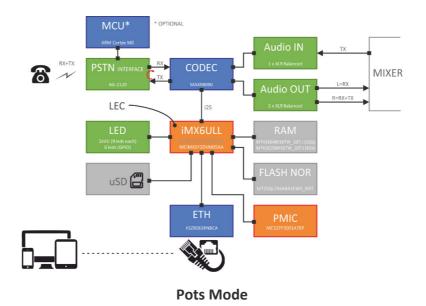
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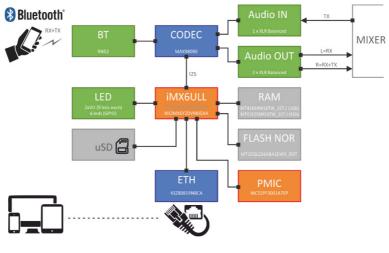
Web Interface - Audio Parameters

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Web Interface - Bluetooth Equalization

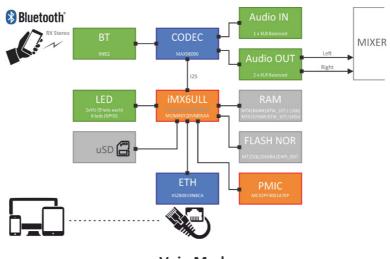




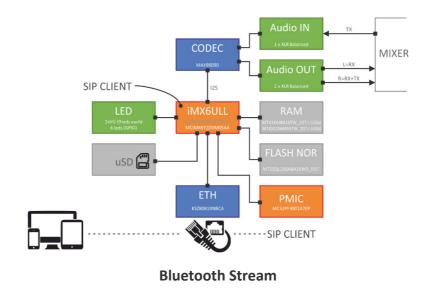


Bluetooth Mode









Technical Specification

GENERAL

Power supply:

90-260 V AC / 47-63 Hz



GENERAL Power consumption:	10 W
Dimensions WxHxD:	483 x 44,5 x 140 mm (1 rack unit 19")
Weight:	< 2Kg
ANALOG AUDIO INPU	т
Connectors:	Balanced on XLR – EMI Suppression
Input impedance:	50 ΚΩ
Nominal Input Level	Adjustable via software: -9 dBu
(sensitivity):	? +15,0 dBu
Level range:	–20,0 dBu
Input level max.:	+20,0 dBu
CMRR input:	>60 dB (20 Hz 🛛 20 kHz)
DIGITAL AUDIO INPL	JT
(OPTIONAL)	
Connectors:	Balanced on XLR – EMI Suppression



DIGITAL AUDIO INPU (OPTIONAL)	JT
Input impedance:	110 ΚΩ
Format:	AES3/EBU & SPDIF
Sample rate:	32, 44.1, 48, 64, 88.2, 96 KHz
Nominal Input Leve	I
(sensitivity):	From 0,0 dBFs to -24dBFs (0,1dB step)
Level range:	0,0 dBFs 🛛 -36dBFs
ANALOG AUDIO OUTPUT	
Connectors:	Balanced on XLR – EMI Suppression
Output impedance:	47 ΚΩ
Output level:	Adjustable via software: -9dBu 🛛 +15,0 dBu
Level range:	–20,0dBu
Output level max.:	+ 20,0dBu



ANALOG AUDIO OUTPUT	
CMRR output:	>60dB (20Hz 🛛 20kHz)
DIGITAL AUDIO OUTPUT (OPTIONAL)	
Connectors:	Balanced on XLR – EMI Suppression
Input impedance:	110 ΚΩ
Format:	AES3/EBU
Sample rate:	32, 44.1, 48, 64, 88.2, 96 KHz
Nominal Input Level	
(sensitivity):	From 0,0 dBFs to –24dBFs
Level range:	0,0 dBFs 🛛 -36dBFs
POTS INTERFACE	
Connectors:	2x RJ11 Line and Telephone set
Output impedance:	Selectable



POTS INTERFA	CE					
Echo cancelle	er > 40dB					
suppression	> 400D					
VOIP INTERFACE						
Version:	SIP Version 2.0 - RFC 3261					
Codec Audio	G.711u (PCM u), G.711a (PCM a),					
Compatibility:	G.722, G.722.1 24/32					
Authentication Methods:	SIP/IAX Dynamic Registration (Register) SIP Static IP authentication					
BLUETOOTH						
INTERFACE						
č	Fully certified Bluetooth version 3.0 audio module, Fully compatible with luetooth version 2.1+EDR Compatible with 1.2 and 1.1 SIP Version 2.0 - RFC 3261					
Connection	Embedded Bluetooth stack profiles:					
mode:	A2DP and HFP/HSP					
IOS:	Supports iAP profile discovery for					



BLUETOOTH INTERFACE

iPhone[®] and iPod[®] Bluetooth accessories G.711u (PCM u)

GENERAL

GPIO Inputs/Outputs: 4 GPI 4 GPO

Communication Port: 1xLAN

Front Panel LEDs: 18+4+2

USB: Type A

Operating Temperature: 0°C ÷ 50°C

COMMUNICATION

Configuration Software: Web Server

Password Protection: Yes

UDP, TCP, HTTP, SNTP. Yes

Supported Network Protocols: HTTP, UDP, TCP, NTP,