

Easy Communication

For Next 10 Years









S-Series VoIP PBX Performance and Power





Easy to Install and Manage

Save time with streamlined installation. Manage and maintain the S-Series on the intuitive graphical UI with point-and-click ease.



Modular and Scalable

Customizable with FXS, FXO, GSM, WCDMA, 4G LTE, BRI, E1/T1/PRI modules and is ready to scale up to 500 user and 120 concurrent calls.



Abundant Built-in Features

Increase efficiency and lower cost of ownership with all-in-one features in a single server without extra licensing fees.



Secure IP Communications

Protected from Internet hackers, computer viruses, eavesdrop with built-in Firewall, TLS, SRTP, and HTTPS standards, and more.



Quick IP Phone Provisioning

Automatically provision phones from Cisco, Yealink, Polycom, Vtech, Mitel, Snom, and enjoy the great interoperability.



Remote Management

Manage and control customer-premised S-Series VoIP PBX easily and securely in a centralized platform.



Award-winning Phone System









S412

Users

 20 Users 8 Concurrent Calls 12 FXS Extensions (Max)

Modular Design

- 4 Customizable FXS Ports 4 Customizable FXO/BRI Ports
- Up to 2 GSM/3G/4G Ports

Call Recording

- · Built-in automatic call recording capability
- . Store recording files in TF card (Micro SD card)





S20

Users

 20 Users 10 Concurrent Calls
 4 FXS Extensions (Max)

Modular Design

- 2 Flexible Module Slots
- Up to 4 FXS/FXO Ports
- Up to 4 BRI Ports
- Up to 1 GSM/3G/4G Channel

Call Recording

- . Built-in automatic call recording capability
- . Store recordings files in TF card (Micro SD card)



im i mini

S50

Users

25 Concurrent Calls
 8 FXS Extensions (Max)

Modular Design

- 4 Flexible Module Slots
- · Up to 8 FXS/FXO Ports
- Up to 8 BRI Ports
- Up to 4 GSM/3G/4G Channels

Call Recording

- · Built-in automatic call recording capability
- . Store recording files in SD card

S100

Users

- 100 Users (up to 200) 30 Concurrent Calls (expandable to 60)
- 16 FXS Extensions (Max)

Modular Design

- . Support 2 Expansion Boards
- Up to 16 BRI Ports
- Up to 2 E1/T1/PRI Ports
- . Up to 16 FXS/FXO Ports
- Up to 6 GSM/3G/4G Channels

Call Recording

- . Built-in automatic call recording capability
- . Store recording files in SD card or USB









D30 Module

EX08 Board

EX30 Board

S300

Users

- 300 Users (up to 500)
- . 60 Concurrent Calls (expandable to 120)
- 24 FXS Extensions (Max)

Modular Design

- · Support 3 Expansion Boards
- Up to 24 BRI Ports
- Up to 3 E1/T1/PRI Ports
- Up to 24 FXS/FXO Ports
- Up to 6 GSM/3G/4G Channels





##**#**



D30 Module

EX08 Board

EX30 Board



- · Built-in automatic call recording capability
- · Store recording files in SD card, USB, or HDD

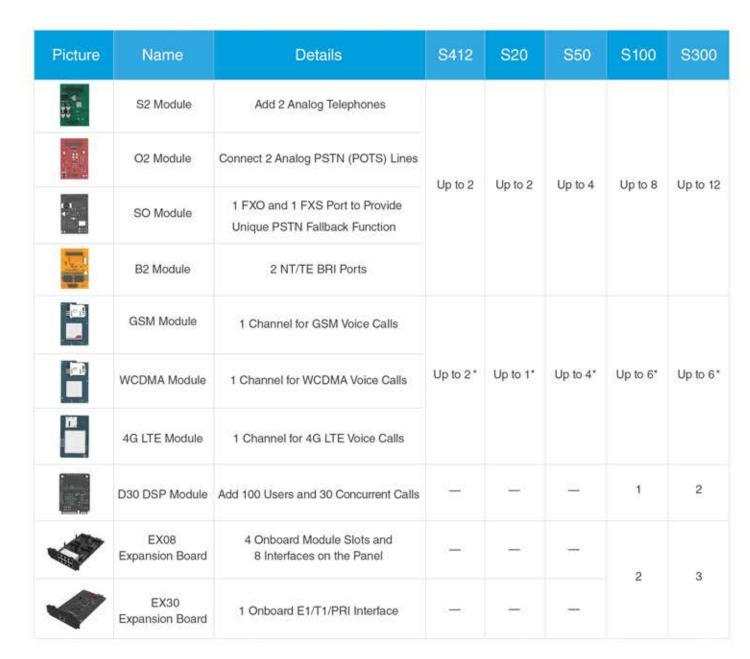
Scalable and Modular Design

Add Telephony Interfaces

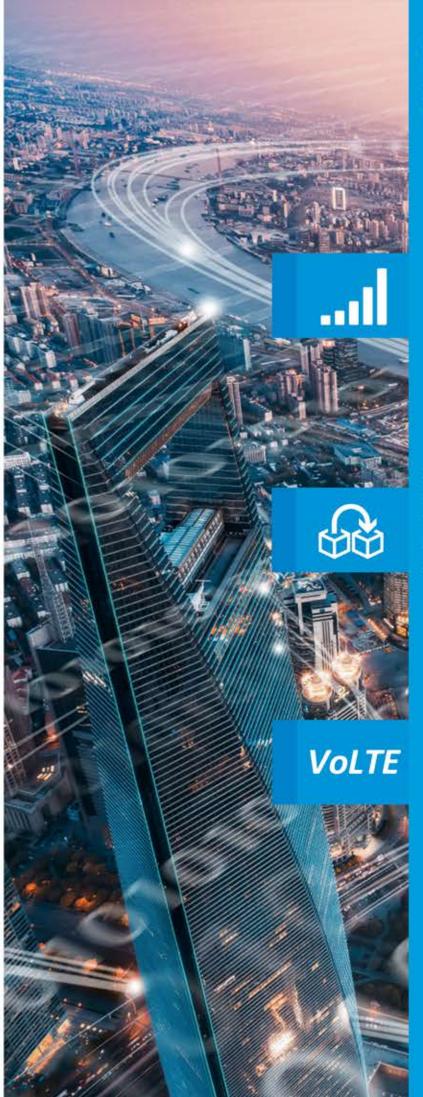
With Yeastar S-Series, all the telephony interfaces are modular and customizable with Telephony Modules and Expansion Boards.

Expand System Capacity

A future-proof investment, Yeastar S-Series allows you to add more users and concurrent calls when business grows.



Only one 4G LTE port could use 4G data.



SIP over 4G LTE on S-Series VoIP PBX

Enable 4G cellular data on SIM card to use 4G LTE as the primary internet connection and enjoy all the powerful features of S-Series VoIP PBX. It is suitable for temporary locations, mobile offices, and places where a fixed-line broadband is not available.

Automatically reroute your internet traffic to the costeffective 4G LTE wireless network and provide businesses with an essential internet continuity option when the primary internet connection fails. With 4G LTE failover, SIP trunks are even more resilient.

Place the phone call over LTE connection rather than the existing 2G/3G voice networks. With VoLTE*, you can expect faster connection time, high-definition call quality. Improving call stability, voice quality, and message integrations, VoLTE offers great benefits to companies.

lease check with your carrier to see If they support VoLTE

S-Series VolP PBX **ITSP Configuration Template More than Compatible**

peoplefone Yeastar conducts in-depth 50+ ITSP Globally interoperability test to ensure **Deutsche Telefon** compatibility with SIP Trunkings around the globe easybell and supports pre-configured templates that include basic flowroute: trunk information and make Proven Interoperability configuration easy and quick. VoiceHost Easy to Configure

IP Phone Auto Provisioning

Yeastar S-Series VoIP PBX allows users to batch configure user information, local phone book, firmware, etc. for a bunch of IP phones at one time, saving time substantially.



150+ Phone Models





Quick Provisioning









Your Choice of Phones



Panasonic





APP Center

Add and Manage New Features Independently





Auto Provisioning

Auto Provisioning is used to provision IP phones in bulk, including all user information, local phone book, firmware, and so on. Phones supported include Yealink, Snom, VTech, Gigaset, Cisco, Polycom, Mitel, Fanvil, Grandstream, Htek.



LDAP Server

LDAP Server provides centralized phone book management, which makes phone book management easy, feature rich and even automated. Once LDAP is set up, you can search the LDAP directory and look up contacts on your IP phone.



VPN Server

The VPN server application provides a way to connect PBX to a remote network in a secure way. The traffic between the PBX and remote network is encrypted, which improves the security of PBX.



Hotel

Hotel App empowers S-Series users to intuitively manage the booking and check-in and check-out of guests, check status of each room, offer mini-bar service, and run personalized billing reports, and more tasks in hotels.



Billing

Use Billing App to allocate call credit and top up extensions, and conduct call analysis. Both prepaid and postpaid payments are supported. Rate can be set according to extensions, time periods, call duration, prefix number and number length.



Conference Panel

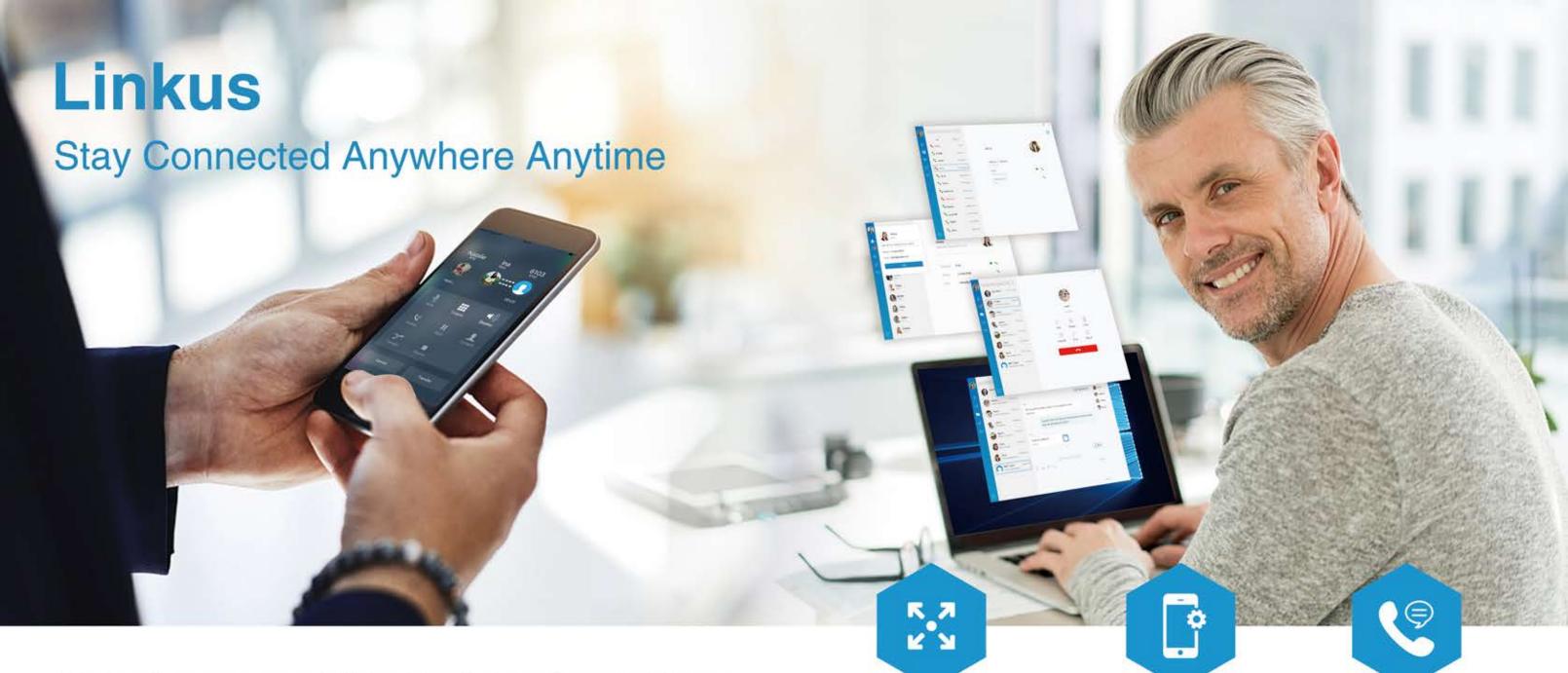
Conference Panel is a visual control panel for your conference calls. You can batch invite people with the dial-out feature in the panel or use your telephone. You can also save all the attendees contact information to the "Contact Group".

QM QueueMetrics Integration

QueueMetrics Integration provides the interface to connect S-Series with QueueMetrics Live or QueueMetrics On-Premise server. The integration helps you to easily generate the queue report daily, weekly, and monthly.

Char utile h+ Integration

This integration provides the interface to connect char, the PMS middleware, which realises the billing, wakeup -call feature engaged with Hotel PMS. As a result, the Hotel PMS and S-Series VoIP PBX are interconnected.



Linkus is a unified communications app for Windows desktop, Mac desktop, iPhone and Android phones.

Designed for Yeastar S-Series VoIP PBX and Yeastar Cloud PBX, it transforms your desktop and mobile phone into a fully-featured office extension, provides convenient call experience, and offers powerful collaboration features like presence, instant messaging, CRM integration, etc.

Feature Highlights

- Make and receive calls anywhere anytime
- · All the office extension features included
- · Start a chat or group chat with instant messaging
- View the presence status of your colleagues
- Personalize your presence description
- Outlook Contacts and Google Contacts integration

Bring Your Extension with You

Make and receive enterprise VoIP calls on your computer and mobile phones while enjoying all the office extension features. Also benefit from true one number reach which keeps your personal numbers private and have instant access to corporate and personal contacts.

Manage and Control Your Calls

Transfer, hold, mute, and record a call to any phone number or extension on the intuitive user interface. You can also see missed calls, check contact histories, listen to voicemails, one-touch recordings, and set up ring strategy and call settings easily on Linkus.

Consolidated Collaboration

Extension list provides an overview of your corporate directory and displays your colleague's presence status. With just a few clicks, you can also initiate a chat or group chat. Seamlessly communicate with your CRM Contacts and enjoy call-popup and call journal on Linkus Desktop Client.

S-Series Specifications and Features

Item	Models	S412	S20	S50	S100	S300			
	Users	8 (SIP+IAX)	20	50	100 (up to 200)	300 (up to 500)			
Capacity	Max Concurrent Calls	8	10	25	30 (up to 60)	60 (up to 120*)			
	Max Analog Ports	12	4	8	16	24			
	Max BRI Ports	4	4	8	16	24			
	Max Cellular Ports	2	1	4	6	6			
	Max E1/T1/J1 Ports	-	-	-	2	3			
Network	LAN	1 10/1	100 Mbps	1 10/100/1000 Mbps					
	WAN	1 10/1	100 Mbps	1 10/100/1000 Mbps					
	IP Services	s	NAT, STUN						
	Protocol			SIP (RFC3261), IAX2					
	VoIP Trunks	4	20	50	100	100			
	Transport Protocol			UDP, TCP, TLS, SR	TP				
VoIP	Audio Codec	G711 (alaw/ulaw), G722, G726, G729A, GSM, Speex, ADPCM, iLBC							
	Video Codec	H263, H263P, H264, MPEG4							
	DTMF	In-band, RFC4733 (RFC2833), SIP INFO							
Automatic Call Recording		Yes	Yes	Yes	Yes	Yes			
IVR		20	32	32	64	128			
Voicemail		5000 min (expandable)		10000 min (expandable)		andable)			
Hot Standby		Yes	Yes	Yes	Yes	Yes			
Firewall		Yes	Yes	Yes Yes		Yes			
T.38 Fax		Yes	Yes	Yes	Yes	Yes			
SMS to M	ail/Mail to SMS	Yes	Yes	Yes	Yes	Yes			
App Cent	er	Yes	Yes	Yes	Yes	Yes			
External S	Storage	TF Car	d (Micro SD Card)		SD Card	SD Card and 2.5 inch SATA HDD			
USB		-	-	1		1			
	Power	DC 12V 3.33A	DC 12V 1A	AC 100-240V 50/60Hz 0.6A max	AC 100-240V 50/60Hz 1.8A m				
Physical	Dimensions (L x W x H) (mm)	290 × 180 × 33	160 x 160 x 30	340 x 210 x 44	440 x 252 x 44	440 x 252 x 44			
	Weight	0.68 kg	0.3 kg	1.48 kg	2.5 kg	2.6 kg			
	Power Consumption	9-25.6 W	1.8-10.6 W	2.1-18.5 W	5.5-41.2 W	6,2-59.1 W			
	Environment	Operation Range: 0°C to 40°C, 32°F to 104°F; Storage Range: -20°C to 65°C, -4°F to 149°F; Humidity: 10-90% non-condensing							
	Mounting	Desktop	Desktop & Wall-mount		Rack-mount				
Features	Automated Attendant (I' Attended Transfer AutoCLIP Blind Transfer Blacklist/Whitelist Call Back Call Detail Records (CD Call Forwarding Call Monitor Call Parking Call Pickup	Call W Caller Confer Custor Dial by DR) DISA Distinct DNIS	aiting ID ence n Prompt Name tive Ringtone t Disturb (DND)	Fax to Email Follow Me Paging/Intercom LDAP Mobility Extension Multi-language System Prompt Multi-language Web GUI Music on Hold One Touch Recording Queue PIN List		Real-time Status Monitor Ring Group SIP Forking Skype Integration Speed Dial Time Condition User Portal User Permission Video Calls Voicemail Voicemail			

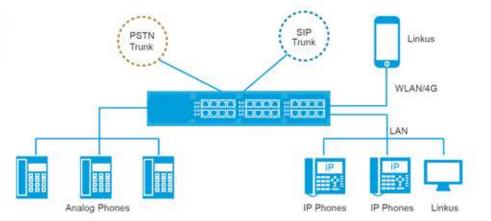
^{&#}x27;If automatic call recording is enabled, the max concurrent calls will be 90

S-Series Meets Your Business Needs

VolP Solution For SMBs

Solution Highlights

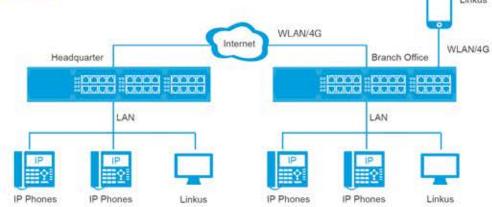
- Linkus provides anywhere anytime connectivity, IM & Presence
- Deploy your choice of IP phones with quick phone provisioning
- Highly reliable, scalable, and secure VoIP solution



Multi-Site & Branch Offices

Solution Highlights

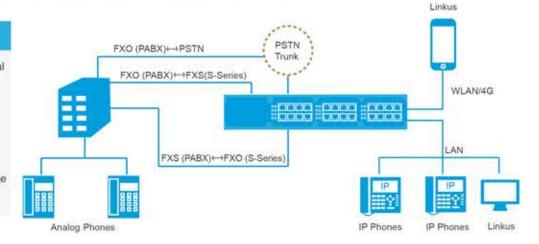
- Secure VPN tunneling between headquarter and branch office
- Linkus provides anywhere anytime connectivity, IM & Presence
- Easy administration, reliable performance and robust features



Combine S-Series With Existing Traditional PBX

Solution Highlights

- Enable the migration with minimal investment and disruption
- Leverage SIP and maintain the continuity of legacy equipment
- Linkus provides anywhere anytime connectivity, IM & Presence





Analog VoIP Gateways **TA Series**



Yeastar TA Analog VoIP Gateways are cutting-edge products that connect legacy telephones, faxmachines and PBX systems with IP telephony networks and IP-based PBX systems. Featuring rich functionalities and easy configuration, TA is ideal for small and medium enterprises that wish to integrate a traditional phone system into IP-based system.

- 4/8/16/24/32 FXS ports or 4/8/16 FXO ports
- Fully compliant with SIP
- Flexible calling rules
- Reliable fax performance with T.38
- Provides high-quality voice compression with industry standard codecs
- Line echo cancellation for 8, 16, 32, 64 or 128 ms echo delays
- · Web-based GUI for easy configuration and management
- Excellent interoperability with a wide range of legacy andIP equipment

Models	TA400	TA800	TA1600	TA2400	TA3200	TA410	TA810	TA1610
Ports	4 FXS	8 FXS	16 FXS	24 FXS	32 FXS	4 FXO	8 FXO	16 FXO
Protocol	SIP UDP, TCP, TLS, SRTP G.711 (alaw/ulaw), G.722, G.723, G.726, G.729A/B, iLBC, GSM, ADPCM							
Transport								
Codec					М			
DTMF	RFC2833, SIP Info, In-band ITU-T G.168 LEC Echo Cancellation, Dynamic Jitter Buffer, VAD, CNG, PLC 1 10/100 Mbps Ethernet							
Voice Capability								
Ethernet								

VoIP GSM Gateways
GSM/3G/4G Channels
TG Series



Yeastar TG Series connect GSM/3G/4G network to VoIP network directly and support two-way communication: GSM/3G/4G to VoIP and VoIP to GSM/3G/4G. It is the best solution ever to connect IP-based telephone systems, and softswithces to GSM/3G/4G network; and also the best fallback solution when landline goes down.

- Link up PBX systems with GSM or 3G WCDMA or 4G LTE networks
- Send and receive SMS messages via Web interface and support bulk SMS
- Save phone bills tremendously with mobile-to-mobile calls and SIP trunks
- · Everything can be easily set up in the Web based management portal
- · Work as a cost-effective backup when the landline network goes down
- · Easy to to integrate: compatibility with mainstream IP-PBX and softswitch

Models Channels	TG100	TG200	TG400	TG800	TG1600
Items	1	2	4	8	16
Protocol	SIP				
Transport	UDP, TCP, TLS, SRTP				
GSM Frequency	850/900/1800/1900MHz				
WCDMA Frequency	850/1900MHz, 850/2100MHz, 900/2100MHz				
Codec	G.711 (alaw/ulaw), G.722, G.726, G.729A, GSM, ADPCM, Speex				
Ethernet	1 10/100 Mbps Ethernet				

E1/T1/PRI VoIP Gateways TE Series



Yeastar TE100/TE200 is a single port or dual port VoIP E1/T1/J1 gateway (VoIP to E1/T1/J1, and E1/T1/J1 to VoIP) that supports up to 30 or 60 concurrent calls. Yeastar TE Series offers SMBs cost-effective additions to a legacy telephone system to bring the true benefits of VoIP.

- Software configurable E1/T1/J1 ports, support PRI, MFC R2, SS7, E&M
- · Cost-effective and flexible call routing
- · Fulfill various needs and cut communication cost
- · Easy-to-navigate Web GUI and simple management
- . Compatible with various ISDN PBX, IP-PBX, and softswitch
- · Stable with advanced hardware and software architecture
- · Low power consumption for your green office

Ports Models	TE100	TE200 2 Ports			
Items	1 Port				
Protocol	SIP				
Transport	UDP, TCP, TLS, SRTP				
Codec	G.711 (alaw/ulaw), G.722, G.726, G.729A, GSM, ADPCM, Speex				
DTMF	RFC2833, SIP Info, In-band				
Voice Capability	ITU-T G.168 LEC Echo Cancellation, Dynamic Jitter Buffer				
Ethernet	Dual 10/100BASE-T Ethernet	Dual 10/100/1000BASE-T Ethernet			

BRI VoIP Gateways TB Series



Yeastar TB200/400 is a compact and reliable standalone BRI VoIP gateway (BRI-VoIP/VoIP-BRI) offering 2 or 4 BRI ports for companies using ISDN BRI lines an easy, cost-effective and flexible integration into any VoIP system or enabling any IP PBX to be connected to the public ISDN network.

- · Software configurable TE/NT modes for easy integration
- · ISDN PBX has access to VoIP network
- Additional ISDN BRI trunking for IP-PBX
- · Cost savings on phone calls via VoIP
- Preserve investment on PBX infrastructure
- · Compatible with various ISDN PBX, IP-PBX, and softswitch

Protocol SIP Transport UDP, TCP, TLS, SRTP Codec G.711 (alaw/ulaw), G.723, G.726, G.729A, GSM, iLBC, ADPCM RFC2833, SIP Info, In-band	Models Ports	TB200	TB400 4 BRI Ports		
Transport UDP, TCP, TLS, SRTP Codec G.711 (alaw/ulaw), G.723, G.726, G.729A, GSM, iLBC, ADPCM	Items	2 BRI Ports			
Codec G.711 (alaw/ulaw), G.723, G.726, G.729A, GSM, iLBC, ADPCM	Protocol	SIP			
	Transport	UDP, TCP, TLS, SRTP			
DTMF RFC2833, SIP Info, In-band	Codec	G.711 (alaw/ulaw), G.723, G.726, G.729A, GSM, iLBC, ADPCM			
	DTMF	RFC2833, SIP Info, In-band			
Voice Capability ITU-T G.168 LEC Echo Cancellation, Dynamic Jitter Buffer	Voice Capability	ITU-T G.168 LEC Echo Cancellation, Dynamic Jitter Buffer			
Ethernet 1 10/100 Mbps Ethernet	Ethernet	1 10/100 Mbps Ethernet			