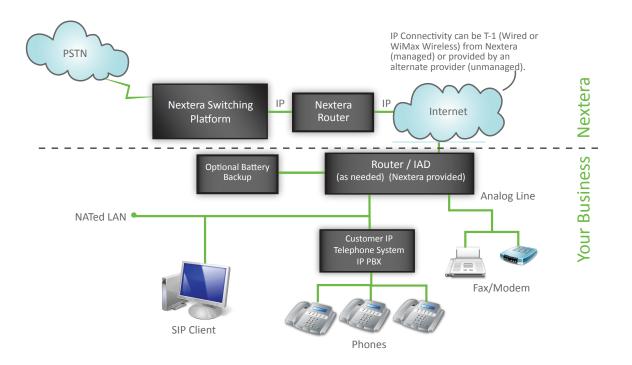


SIP Trunking

Nextera's SIP Trunking delivers IP access and essential telephone features to customers with deployed IP PBXs and IP phones. SIP Trunking configurations are established with a customer-specified amount of voice channels (simultaneous voice calls). Combine SIP Trunking with Internet access from Nextera (i.e. T-1 (Wired or WiMax Wireless) to provide a complete Nextera-managed service.



Codec

The number of simultaneous voice calls available is based upon two factors: 1.) the amount of available bandwidth (i.e. 1 Wired T-1 provides 1.5Mbps of available bandwidth), and 2.) the voice codec selected by the customer – G.711 or G.729

Voice Codec Used	Bandwidth Used Per Call	Call Quality	Supports Fax/Modem*
G.711	95.2K	Excellent	Varies
G.729	39.2K	Good	No

G.711 codec provides uncompressed high quality voice and generally supports analog service such as fax, modems, alarm lines and credit card machines.

G.729 codec provides compressed voice with less fidelity than G.711 codec normal analog phone lines. G.729 does not support fax, modems, alarm lines or credit card machines. Nextera will automatically adjust to G.711 in the event that fax and modem tones are detected, however, is best effort for fax, modem, alarm or other anlalog devices when using G.729.

SIP Trunking Overview

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SIP Trunking continued

The chart below identifies the maximum number of calls (voice channels) supported based upon the available bandwidth.

T-1 Wired

Number of T-1s	Available Bandwidth	Max # of Voice Channels using G.711*	Max # of Voice Channels using G.729*
1	1.5M	16	39
2	3M	32	78
3	4.5M	40	117

* Due to limitation of router, max number of T-1's available is 4.

T-1 WiMax (Wireless)

Available Bandwidth	Max # of Voice Channels using G.711*	Max # of Voice Channels using G.729*
1.5M	16	39
2M	21	39
3M	32	39
4M	39	39
5M	39	39
6M	39	39

Why Choose Nextera's SIP Trunking?

 Value Capitalize on your equipment investment — an IP PBX with SIP Trunking is a native IP interface. A single service that handles both your voice and internet needs. Dynamically allocates bandwidth for internet usage as voice calls connect and discon- nect, providing the ultimate value. Efficiency Automatically allo- cates inbound and outbound data and voice traffic based on your business needs. Identifies each DID to receive inbound calls only, place only outbound calls, or a combination, allow- ing you to reduce the number of analog line and increase opera- tional efficiency. 	 additional channels and/or bandwidth. More than double voice capacity by utilizing compression codecs. 	Spend less time rout- ing calls by assign- ing phone numbers (DIDs) to each em- ployee, fax machine, and modem. Create internal ef- ficiencies to ulti- mately save time and money. Converged access with managed ser- vice allows dynamic bandwidth utiliza- tion.	 Service Provides incoming caller ID so you know who is calling. When you are prepared to provide the service your customer de- serves, the result is improved customer satisfaction. If managed service, T-1 Failsafe automati- cally re-routes calls to a pre-selected number if T-1 is un- available.
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Customer's Telephone System

The customer's telephone system must be IP enabled to accommodate SIP Trunking and must be configured to support the desired number of voice calls.

Nextera's SIP Trunking includes:

/29 Subnet with 8 IP addresses (6 usable) • 10 email boxes • Caller ID • Hunting • Local and EAS Calling • Call Blocking (O, 1+, 411, International, Local) • Porting or Assignment of Telephone Numbers • Account Codes (non-validated) • Inbound Only Calling Capability • Outbound Only Calling Capability • Combination Inbound/Outbound Calling Capability

Optional Features: Account Codes (validated) • T-1 Failsafe (forwards calls to a customer-selected TN when T-1 service is unavailable) if managed service • VPN (Virtual Phone Numbers - Long Distance Telephone Numbers from remote locations which ring to a customer-identified DID. Incoming calls only.) • OnNet Remote numbers available with dedicated T-1

We deliver the services you need in today's competitive business environment! Contact us today at 877-639-8372 or www.nextera.net

Nextera Communications