IP Telephony Guide

This document provides an introduction to IP calls with TX3 voice entry products (for instance the TX3 Nano, TX3 Touch, TX3 InSuite, and TX3 MiEntry).

SIP

SIP (Session Initiation Protocol) is a protocol that controls multimedia messaging on an IP network. TX3 voice entry products use SIP for IP communication. To make audio and video calls using SIP, you need at least 2 SIP clients and a SIP server.

SIP clients

SIP clients are devices that communicate with each other using SIP. The TX3 Nano, TX3 Touch, TX3 InSuite, and MiEntry app are SIP clients.

Some SIP clients can send video, some can receive video, and some can both send and receive video. Two-way video calls are possible only if both clients can send and receive video, as shown in Figure 1.

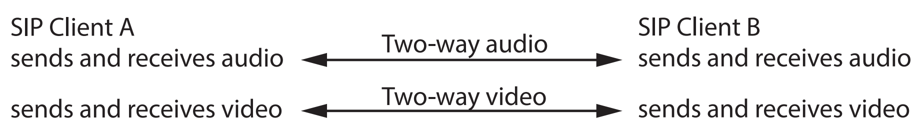

Figure 1. Two-way video call

If one client can send and receive video, and the other can send but not receive video, then the video call is one-way, as shown in Figure 2. The TX3 Nano and TX3 Touch send video but do not receive it. Therefore, a call from the TX3 Nano or TX3 Touch is a two-way audio, one-way video call.

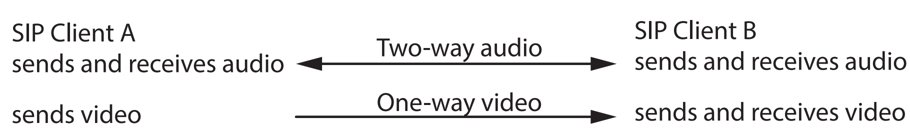

Figure 2. Two-way audio, one-way video call

SIP server

The SIP server is a computer or program that monitors and establishes the call between the SIP clients. The SIP clients must be able to access the SIP server.

Three kinds of SIP servers

Note:	See the manual for the specific TX3 product for information on the specific SIP services supported for that product.

There are three options for managing a SIP server:

- Local SIP server (for example Kamailio)
- Remote SIP server provided by a third party (for example OnSIP)
- Customer-installed SIP server on a cloud service (for example Amazon Web Services)

If the SIP server is local, then in most cases, SIP clients from outside the local area network cannot access it. Figure 3 shows a local SIP server that provides service to SIP clients on the local area network. SIP clients that are not on that local area network cannot access the SIP server.

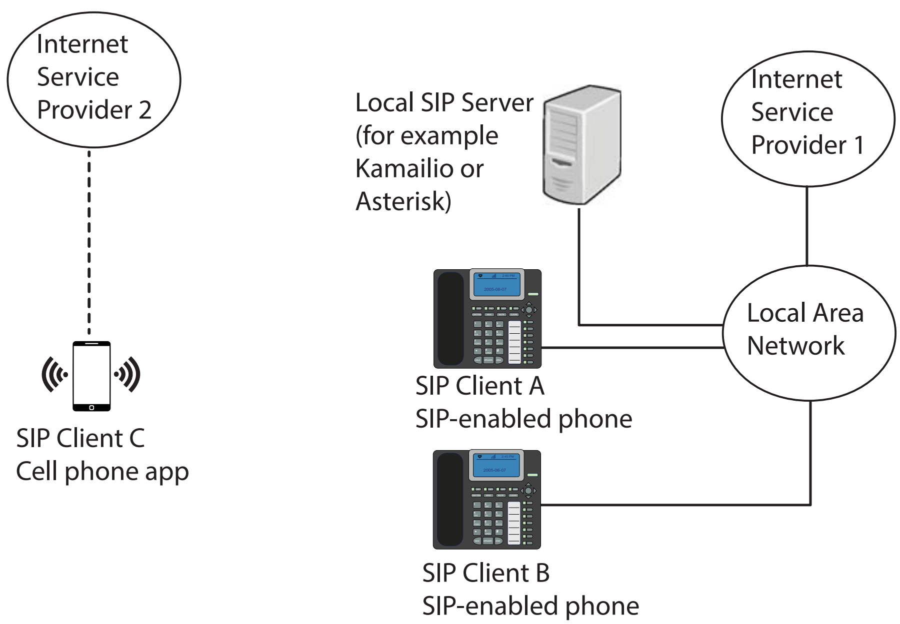

Figure 3. SIP client C cannot access the local SIP server

Third-party SIP servers (for example OnSIP) and SIP servers on a cloud service (for example Amazon Web Services) are globally accessible: they are accessible by all SIP clients no matter how the SIP clients are connected to the Internet. Figure 4 shows a globally accessible SIP server that provides service to SIP clients that are connected to the Internet in different ways.

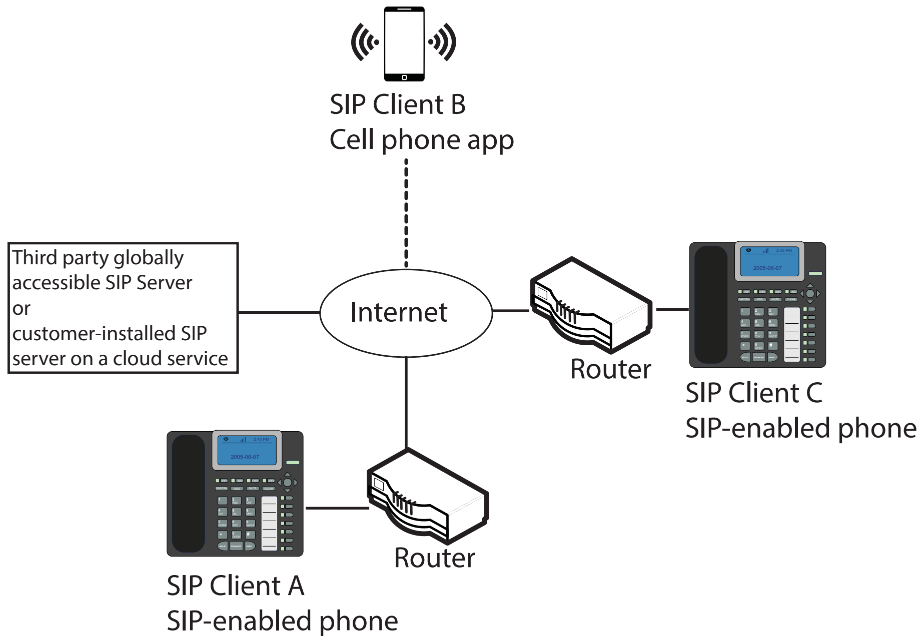

Figure 4. Globally-accessible SIP server

Figure 5 shows an example application with the TX3 Nano and MiEntry as the SIP clients, and OnSIP as the globally accessible SIP server.

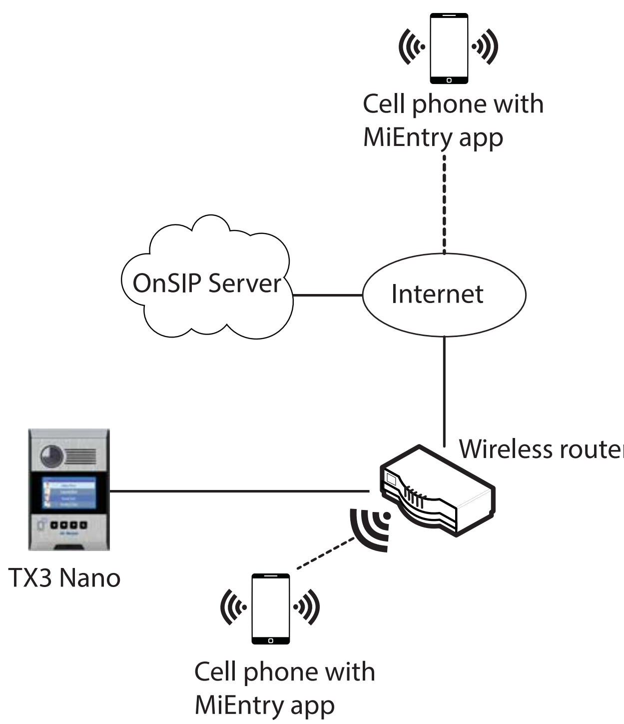

Figure 5. Example application with TX3 Nano and OnSIP

OnSIP

OnSIP (http://onsip.com/) is a company that provides a globally accessible SIP server that supports both audio and video. If you have an OnSIP administrative account, then you can use the TX3 Nano to quickly create SIP accounts for the TX3 Nano and for residents. In this situation, the TX3 Nano and the residents are the SIP clients, and OnSIP is the SIP server. See LT-1194 TX3 Nano Configuration Manual for more information.

SIP clients must be registered with the SIP server

SIP clients can communicate with each other only after they are registered with the SIP server. The SIP server registers a SIP client based on the SIP client's **SIP account details**, which usually consist of a username, password, and address of the SIP server. Therefore, each SIP client must be configured with a username, password, and address of the SIP server, and the SIP server must hold a list of usernames and passwords for all the SIP clients.

Figure 6 shows the information that the SIP clients and server must be configured with in order for registration to succeed.

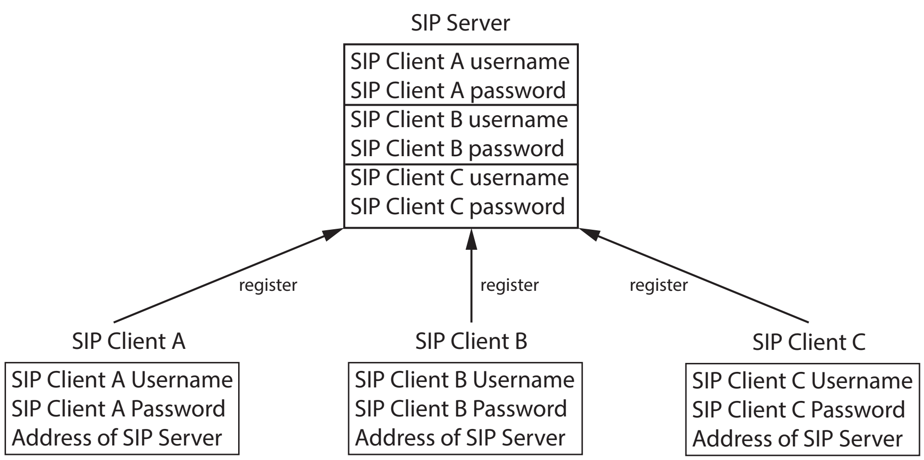

Figure 6. SIP registration

SIP Trunks

A SIP trunk is a method of connecting SIP clients to the traditional telephone network (also call the public switched telephone network or PSTN). If the SIP server has a trunk to the PSTN, then the SIP clients can have traditional phone numbers and make and receive calls from traditional phones.

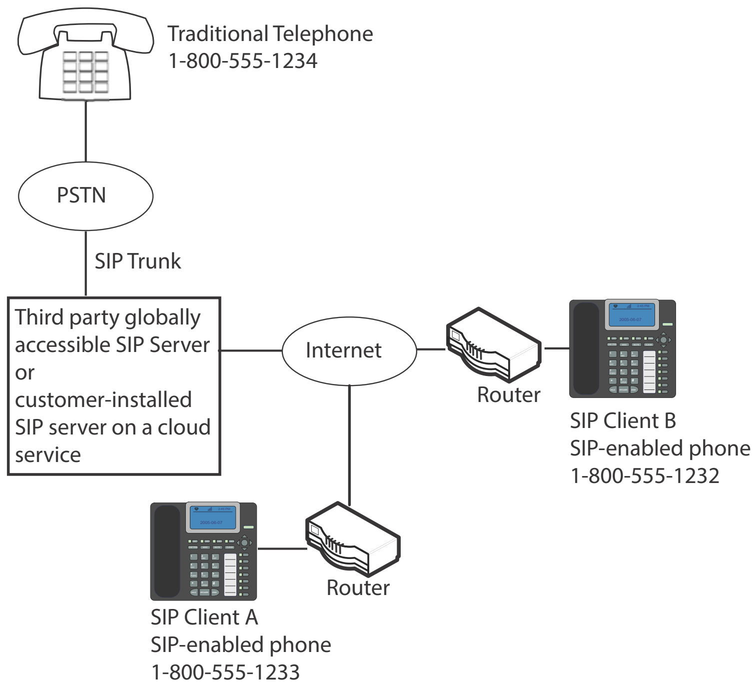

Figure 7. SIP Trunking

The SIP clients and SIP server are in constant communication

In TX3 applications, the SIP clients and SIP server communicate with each other at least every 5 minutes, even if no call is in progress. If there is a break in the connection (for instance, a cable is unplugged then plugged back in), then communication will be restored in 5 minutes.

SIP clients must use the same codecs

SIP clients use codecs to encode and decode audio and video signals. Both SIP clients must use the same codecs in order to communicate with each other.

The table below lists the supported codecs. ASIP client must support the applicable codecs in order to communicate with a TX3 product.

The TX3 MiEntry app for Android and iOS is compatible with all TX3 products.

Product	Audio Codecs	Video Codecs
TX3 Nano	G.711 μ-law	H.263 H.263+ H.264 MPEG4
TX3 Touch	G.722-64k G.722.2 G.722.1-32k G.722.1-24k G.711 A-law-64k	H.263, H.263+
TX3 InSuite	G.711 μ-law-64k G.711 A-law-64k	The TX3 InSuite does not use SIP for video; it uses the video from the ONVIF camera (see LT-6082)

Bandwidth

Residents who receive calls on their cellphones from TX3 products should be aware of the bandwidth that the call uses.

A two-way audio and one-way video call from the TX3 Nano and TX3 Touch uses 1.5 megabits per second. A thirty second call is 4.5 megabits.

Supported SIP Services

See the manual for the specific TX3 product for information on the specific SIP services supported for that product.

These documents are available on http://www.mircom.com.

- LT-995 TX3 Touch Screen Configuration and Administration Manual
- LT-1194 TX3 Nano Configuration Manual
- LT-6082 Unified Building Solution Administration Guide
- LT-2087 Emergency Phone Application Guide

Example Wiring Diagram

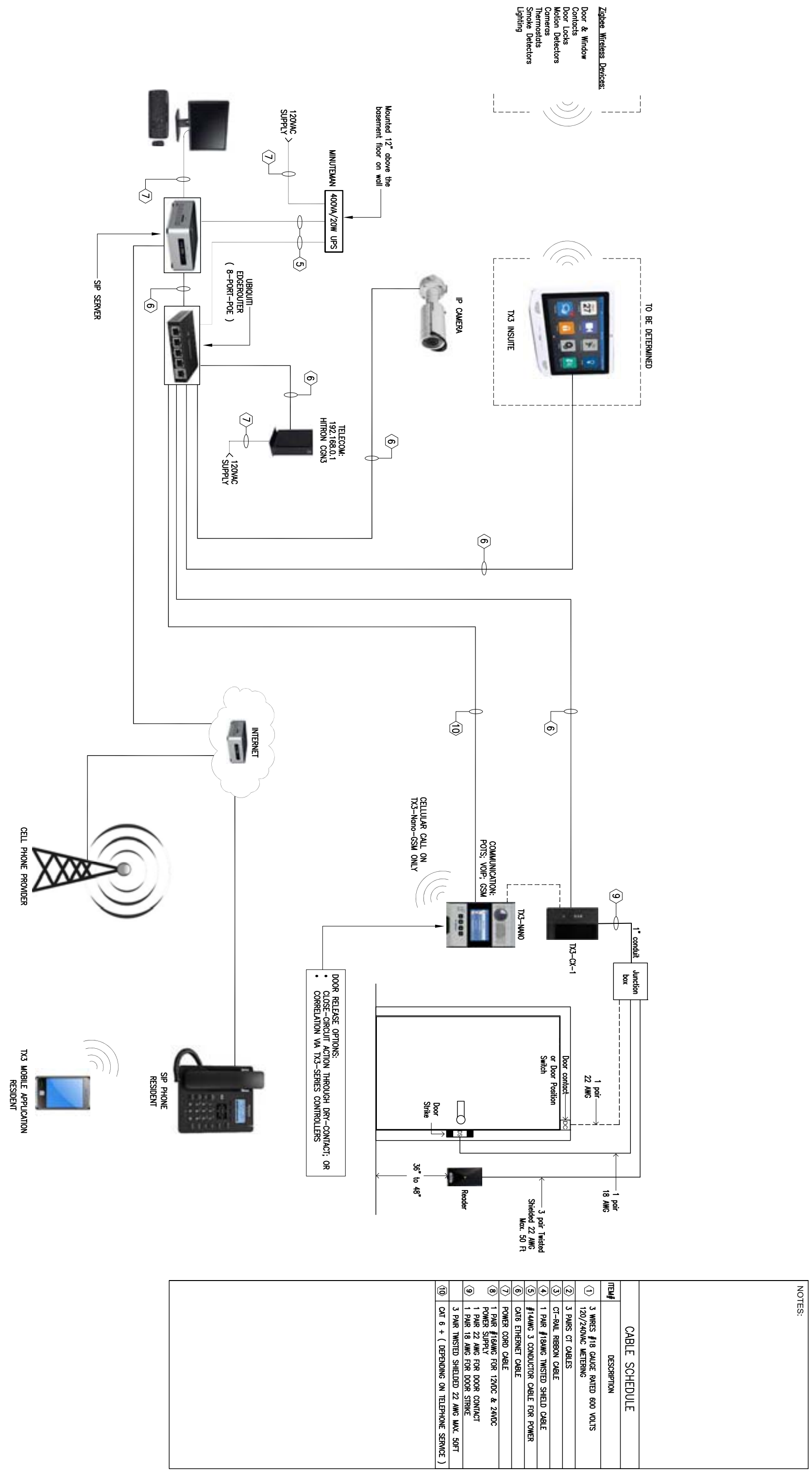

Figure 8. Example Wiring Diagram

Glossary

Codec: A program that codes and decodes audio and video signals.

Device: This refers to any technology that answers a SIP call. It can be a cell phone, tablet, IP phone, computer, or laptop.

DHCP (Dynamic Host Configuration Protocol): DHCP is a method of automatically assigning IP addresses to devices on a network. On a LAN, the router usually has a DHCP server that assigns IP addresses to all devices on the LAN.

IP address: An IP address is a series of four numbers separated by periods (for instance, 10.10.8.2) which identifies a device on an IP network. IP addresses can be local (used within the local network) or global (assigned by the Internet service provider and used on the Internet). Each Master Node on a TX3 network has a local IP address which is used within the TX3 network. The router maps the local IP addresses onto global IP addresses so that the Master Nodes can be accessed over the Internet.

Kamailio: An open-source SIP server for Linux and Unix. (http://www.kamailio.org/).

LAN (Local Area Network): An IP network in a limited area, such as a building, that all the devices are connected to.

MAC address: Each device's network interface has a MAC (media access control) address. This address uniquely identifies the device on the network.

Octet: An octet is the part of an IP address between the periods. For example, the IP address 10.10.8.2 consists of 4 octets: 10, 10, 8, and 2.

OnSIP: A company providing online SIP services (http://onsip.com/).

Port: A port is a number associated with an IP address and used by a specific application. Ports are like telephone extensions. Just as one telephone number can have many extensions, one IP address can have many ports, where each port is used by a different application. For example, the TX3 Configurator uses port 8080 to communicate with Touch Screen Master Nodes, and port 14000 to communicate with non-Touch Screen Master Nodes.

PSTN: Public switched telephone network (the traditional telephone network).

Registered: All devices that use SIP must be registered with the same SIP server.

Router: A router is a device that connects two or more networks together. For example, a router connects a LAN to the Internet.

SIP (Session Initiation Protocol): A protocol for controlling voice and video communication on an IP network.

SIP account details: A SIP username, SIP password, and address of the SIP server.

SIP Client: SIP clients are devices that communicate with each other using SIP.

SIP Server: A computer or program that monitors and establishes the call between SIP clients.

SIP Trunk: A method of connecting SIP clients to the PSTN.

SIP Username (SIP ID): Every SIP client has a unique SIP username (also called SIP ID).

SIP Password: Every SIP client has a password for registering with the SIP server.

Siremis: An open-source web management interface for Kamailio. (http://siremis.asipto.com/)

Subnet mask: A subnet is a way of dividing a network into groups. When the IP addresses of the devices share the first three octets, for instance 128.15.1.x, then the devices are on the same subnet and the subnet mask is 255.255.255.0.

Switch: A switch is a device that connects devices to each other on a network.

TCP/IP: The group of protocols that specify how computers communicate with each other on the Internet.

VOIP: Voice over IP.

WAN (Wide Area Network): A WAN is a group of LANs that are connected over a large area.

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