Luxsys IP-PBX

Luxsys IP-PBX Specifications

1. IP-PBX Basic configurations

Specifications	Product	
Manufactured by	Luxsys Inc., Korea	
Model	IP-50	
Features	Default 100 Extensions (Supports maximum 200 users) Max. 50 concurrent calls	
	Supported protocols: SIP, H323, IAX2, SCCP, MGCP, UNISTIM, SKYPE, etc.	
	By default configured for All-IP. (Can be extended for PRI /TDM)	
	Detailed Specifications	
Product	ALL IP based IP-PBX system that supports Voice recording, ARS/IVR, VMS and etc. features. This model can configured to work with major VOIP protocols such as: SIP, H.323 and IAX2. This model carried one available slot for PRI E1/ FXO/FXS card in order to support PSTN backup feature.	
Configuration	Tested and approved by Telecom in Korea. (Certified) Enterprise CRM with IP-PBX Management Software Fully functional behind NAT, on existing network infrastructure. Built-in Firewall, QoS, Router, DHCP and NTP. Local/External Extensions with support of PBX features. Remote Support and Maintenance The internal storage included for voice recording 19” Rack size, fits standard cabinet as standard network equipment. Best choice for call center with up to 50 concurrent calls 30% Cost efficient over analog PABX	

IP-PBX (Continued)

IP-PBX Standard configuration

Configuration	Specifications
Box Type	19" Small Rack Type
CPU	x86 CPU
Main Board	Intel Board
O/S	Linux
Network	10/100/1000 RJ45
SIP	UDP and TCP Protocols
Voice Recording	Total voice recording is supported by default. (For over 10 concurrent optionally can be installed a NAS storage or harddrive)
CTI	CTI is supplied with CRM software.
CRM	Customer Relationship Management Software with CTI feature.
(Optional) PSTN Expansion card	Supports 1Port, 4Port, 8Port, E1, 2E1, 4E1 Carries DSP Chipset with Hardware Echo Cancellation
Local Extension (Default/IP)	100
PSTN (Max. Channels)	120(4E1)
Extensions (Max./IP)	200
Max. Concurrent Calls	50
Soft Phone	The softphone is not included. 3rd party softphone software can be used with this product.
Size	19" Rack Type, 1.5U
Configuration	Specification

Luxsys IP-PBX Basic Configuration

IP-20 Model

IP-PBX Specifications

Product	Notes
Manufactured by	Luxsys Inc.
Model	IP20
Performance	50 Agents(Maximum 100 Agents)
	Supported Protocols: SIP,H323, IAX2, SCCP, MGCP, UNISTIM, ETC
	FXO/FXS Capable, By default All-IP based IP-PBX
	ARS, VMS, CTI, CRM, Recording

Description	ALL-IP IP PBX designed for SME with the built-in ARS,VMS, Voice recording features. It supports widespread SIP protocol, also this equipment can be configured to work with H.323 and IAX2 protocols. The PSTN backup can be configured by installing optional FXO,FXS or E1 card. The optional analog/digital card can be used to integrate VoIP with analog PABX.
Features	⌖ Complain with ISP and Telecoms in Korea. ⌖ Supplied with GUI CRM and IP-PBX Management software Korean version. ⌖ Supports NAT. ⌖ Built-in Firewall, ⌖ Local/External Extensions, basic PBX features. ⌖ Remote Support and Maintenance ⌖ Configurable for storage to increase capacity for voice recording ⌖ 19” Rack size, fits standard cabinet as standard network equipment. ⌖ Best choice for call center with up to 20 concurrent calls ⌖ 30% Cost efficient over analog PABX

Luxsys IP-PBX

IP20 (Continued)

Product Specifications

Product	Configuration
Box Type	19” Small Rack Type
CPU	X86 CPU
Main Board	Intel Board
O/S	Linux
Network	10/100 RJ45
SIP	UDP and TCP Protocols
Voice Recording	Total voice recording is supported by default. (For over 10 concurrent optionally can be installed a NAS storage or hard drive)
CTI	CTI is included with CRM software.
CRM	Customer Relationship Management Software with CTI feature.
PSTN (Analog/Digital)	Supports 1Port, 4Port, 8Port, E1, 2E1, 4E1 cards Carries DSP Chipset with Hardware Echo Cancellation
Extension (Default/IP)	50
PSTN Channels (Max. Channels)	60(2E1)
Agents(Max./IP)	100
Concurrent Calls	20
Soft Phone	The softphone is not included. 3rd party softphone software can be used with this product.

Product specifications

Basic configuration

Features	Specifications
Default Configuration	Default IP phones lines: 50~200 lines. 8 Analog lines
	8Port FXS/FXO can be installed optionally. PRI E1 card can be configured to support up to 120 channels
	Maximum 55 Analog and VOIP concurrent calls
	Capable to process 3cps (call per second)
	G.711(A/μ-Law) : MOS 4.3(Waveform 6kbs)
	G.729a : MOS 4.0(8kbps)
	Speech Delay : about 20~30ms
	The success call rate is 99.9% over 24 hour call processing.
	Call Forwarding/CF on Busy/CF on No Answer/Roaming can be configured from phone, Web and CRM GUI.
	The dialing time for receiving calls can be adjusted.
	Inbound/Outbound groupings. Route incoming calls to IVR/ARS, Ring groups (ring all group, hunt, round robin, least received, last received, memory hunt).
	Call Waiting feature
	CAC, Call restriction.
	Do Not Disturb feature.
	Music on Hold (Call hold)
	Call pickup
	Blind Transfer
	Attended Transfer
	Conference call
	Callback busy subscriber feature
	Redial / dial last received number
	Specifications

Product specifications (Next)

Basic configuration

Features	Specifications
Call routing and Dial plan	No prefixes for local/outbound/international calls are required.
	Easy to enable/disable PBX Features (same as cell-phone in Korea). *88 Activate CF, *880 Di-activate. Add 0 at the end to disable. *55 to set all features from interactive voice response.
	Black list
	Hide CallerID for outbound call
	ARS (Interactive voice response) can be configured from GUI IVR designer. Three possible way to upload IVR media content: - wav, mp3, gsm files are supported. - TTS (Text-To-Speech) - Record voice from telephone.
	Divert calls from IVR to departments and directly to local extensions.
	Voice Mail
	Time scheduling
	Call center features: Click-to-Dial, Power-Dialer, URL-Dialing, and etc.
	Record VMS response messages
	3 Distinctive bell ringing (Outbound calls, Local extensions, Call back)
	Call parking
	Emergency call
	Specifications

Product specifications

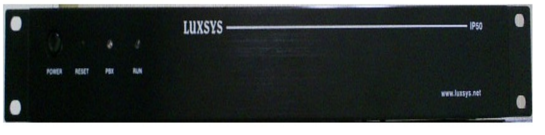
SIP Protocol

Features	Specifications
Supported Protocol Standards	**RFC 3892** (The Session Initiation Protocol (SIP) Referred-By Mechanism (new))
	RFC 3842 (A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP))
	RFC 3665 (SIP Basic Call Flow example)
	RFC 3515 (The Session Initiation Protocol (SIP) Refer Method)
	RFC 3489 (STUN - Simple Traversal of User Datagram Protocol (UDP) Through Network Address Translators (NATs)
	RFC 3428 (Session Initiation Protocol (SIP) Extension for Instant Messaging)
	RFC 3389 (Real-time Transport Protocol (RTP) Payload for Comfort Noise (CN)
	RFC 3361 (Dynamic Host Configuration Protocol (DHCP-for-IPv4) Option for Session Initiation Protocol (SIP) Servers
	RFC 3325 (Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks)
	RFC 3311 (The Session Initiation Protocol (SIP) UPDATE Method)
	RFC 3265 (Session Initiation Protocol (SIP)-Specific Event Notification)
	RFC 3264 (An Offer/Answer Model with the Session Description Protocol (SDP))
	RFC 3263 (Session Initiation Protocol (SIP): Locating SIP Servers)
	RFC 3261 (Session Initiation Protocol)
	RFC 2976 (The SIP INFO Method)
	RFC 2833 (RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals)
	RFC 2617 (HTTP Authentication : Basic and Digest Access Authentication)
	Specifications

2 Structure and view

IP-PBX structure

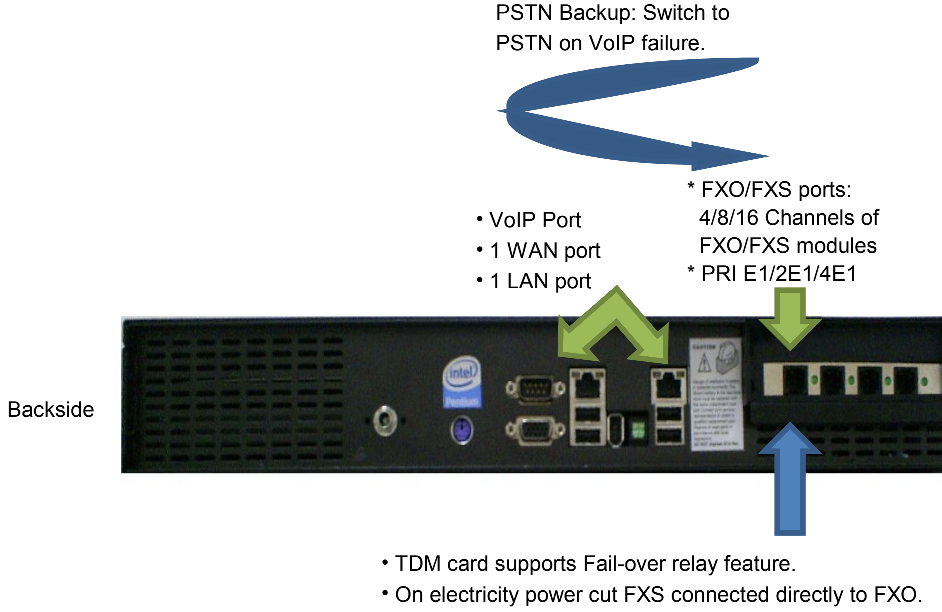
Category	Notes
Manufactured by	Luxsys Inc.
Category	Notes

IP-PBX picture	
Front panel	
Backside	
Installed	

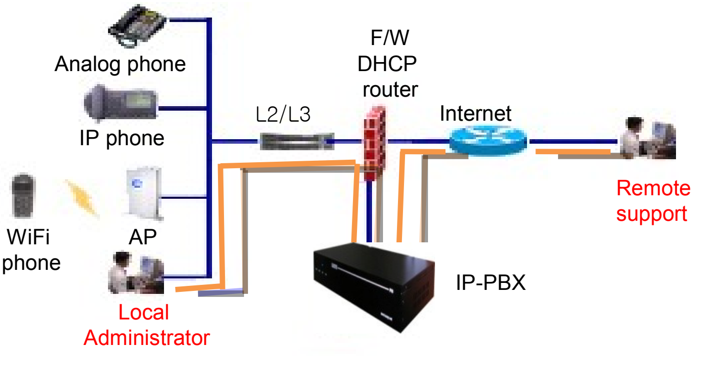
3 Performance

3.1 System structure

Configuration



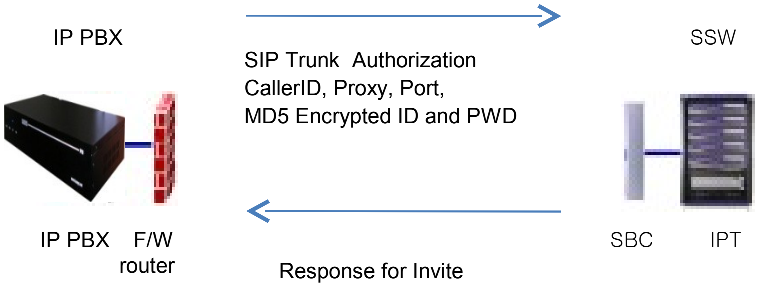
S/W Upgrade



- Software upgrade methods. (Local/Remote Administrator)
  1) Partial upgrade : The IP-PBX is be upgraded by IP-PBX Management software.
  2) System upgrade : System can be upgraded remotely on customer requests.
- After upgrade all settings and configurations are remained same.
- After upgrade it is possible to revert system back to the previous version .

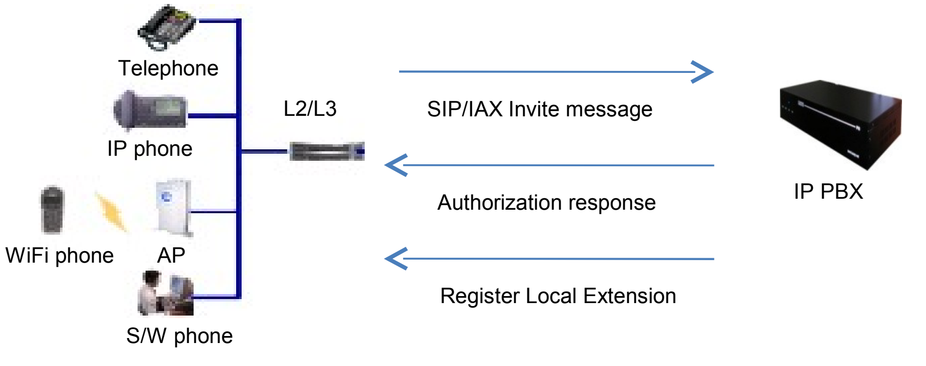
3.2 Installation

SIP Trunk Authorization



- Trunk Authorization requirements:
  1) Centrex: ID, PWD, Domain (Proxy) and Port Number.
  2) Backbone: Proxy Server, Authorized by IP/ IP-PBX Device
  3) IAX trunking is used for communication between IP-PBX devices.

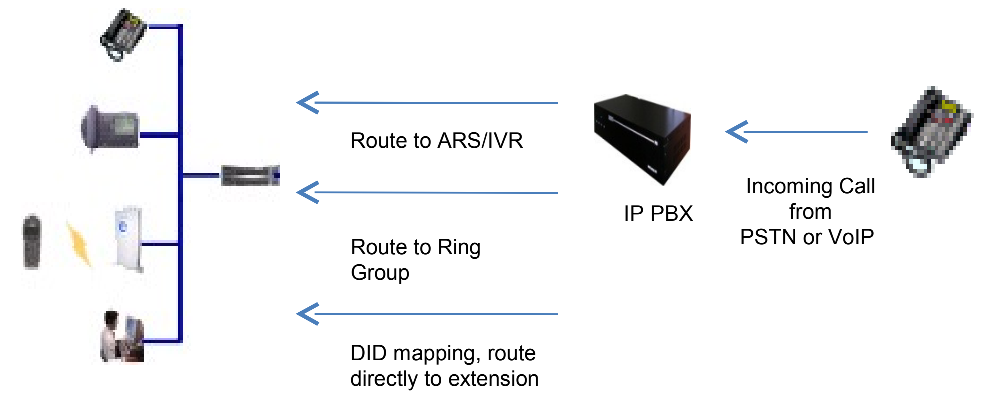
Register Local Extensions



- The devices that support auto-provisioning can be configured and upgraded via TFTP.

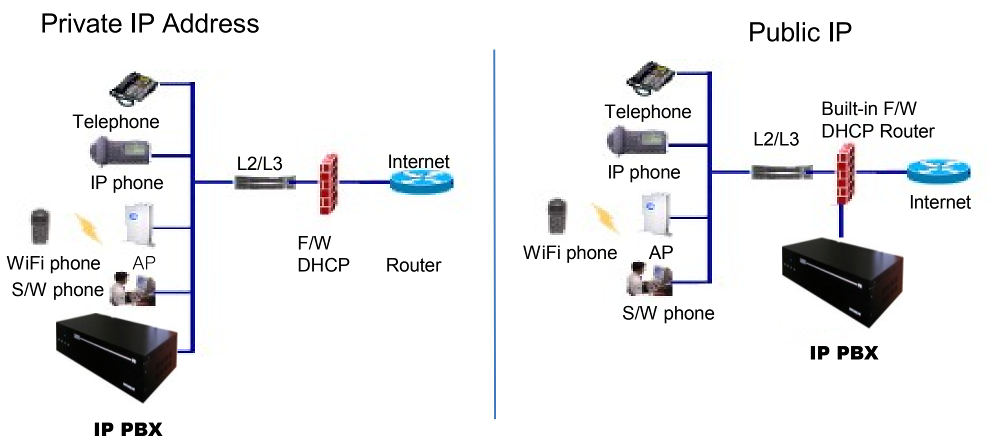
3.2 Installation

Call routing



• All incoming calls can be diverted to ARS/IVR. From interactive voice response callers are routed to ring group. The DID/DOD numbers are mapped directly to extensions.

IP Address (Public, Private) Configuration



• Can be configured for both private and public IP network configuration. Automatically supports and configures NAT, the IP-PBX comes with built-in firewall, DHCP, VLAN and NTP modules.

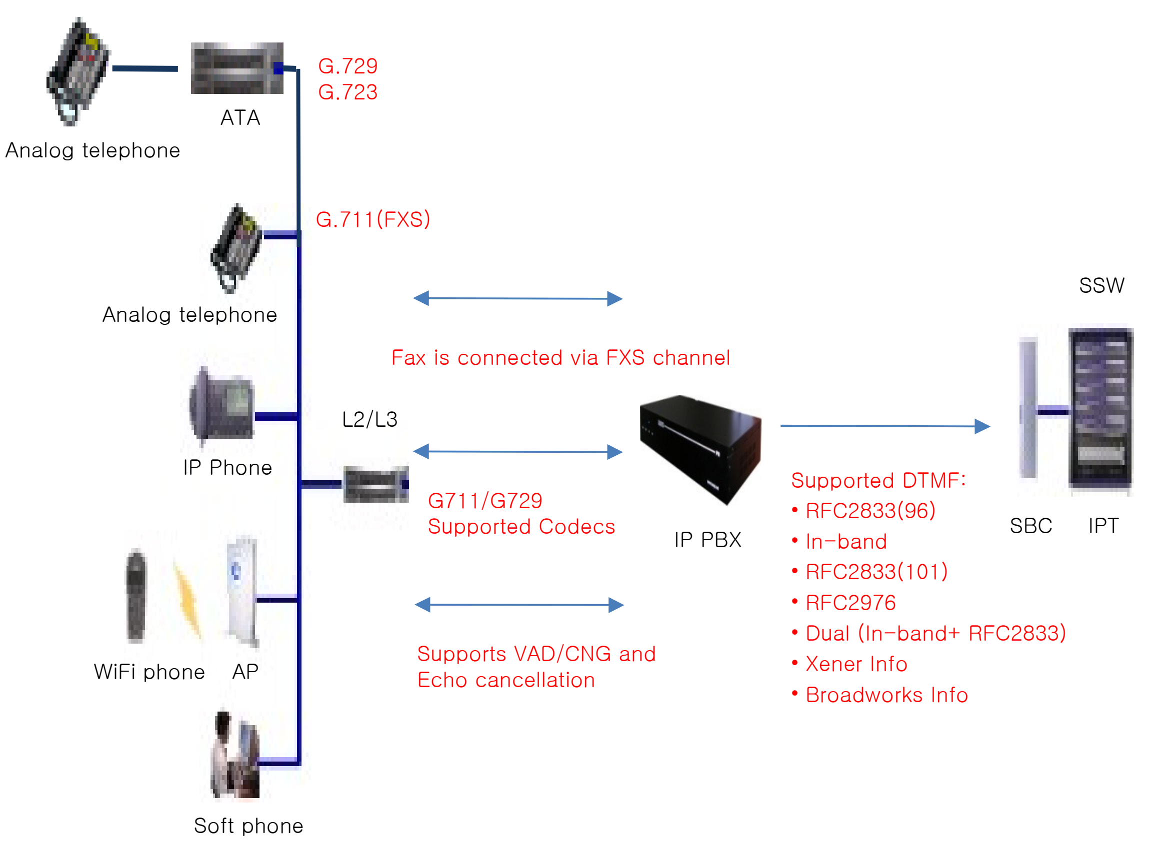
3.4 Supported Devices

IP phones

- IP PBX supports wide range of SIP VoIP devices.
- Certified products for IP-PBX: Dasan TPS, Samsung and Moimstone.
- Cisco, Avaya, Nortel, Polycom and SNOM ip phones are also supported.
- Analog phones are connected to FXS port or via AGW/ATA device.

3.5 Media Processing

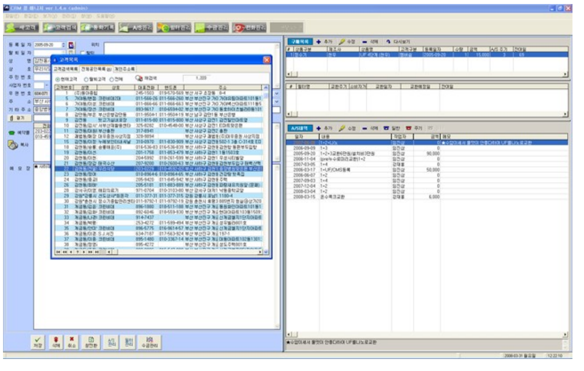
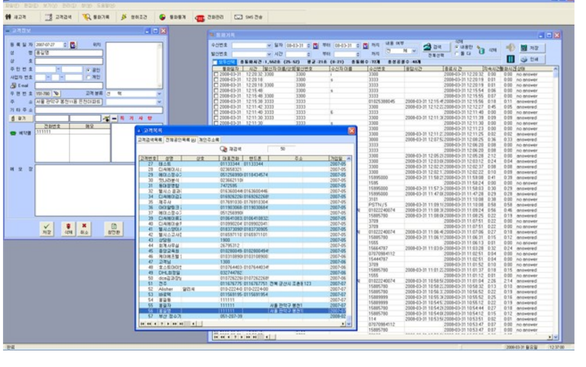
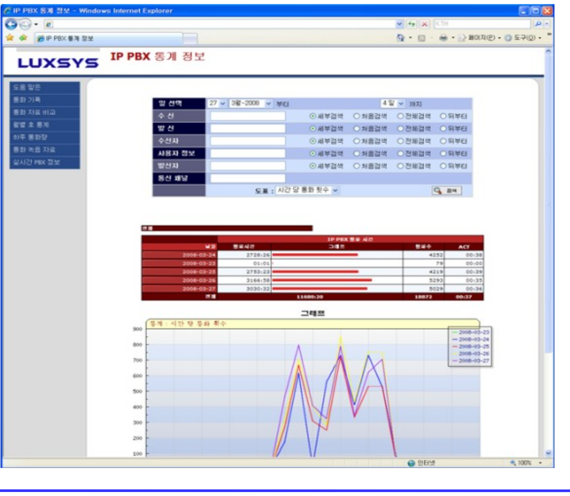
Codec and DTMF modes



- PSTN calls are processed with G711a/u, ATA devices can support G729.
- Voice recordings can be stored in WAV, WAV49, OGG, MP3 or GSM file formats.
- Codec order can be configured from Call Manager software.
- Supports codec auto-negotiation.
- Supported DTMF modes: In-band, RFC 2833, RFC2976 and Info.
- Supported FAX protocols: T.30/T.37/T.38. Possible to connect Fax devices to FXS module, IP-PBX can be configured to operate as FAX-to-Email.

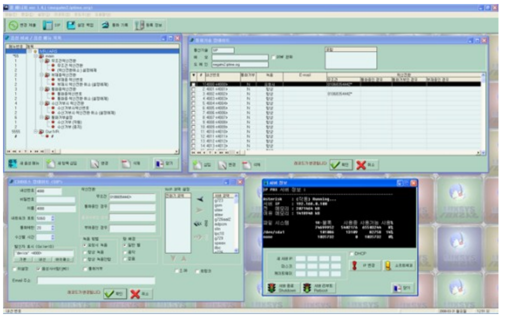
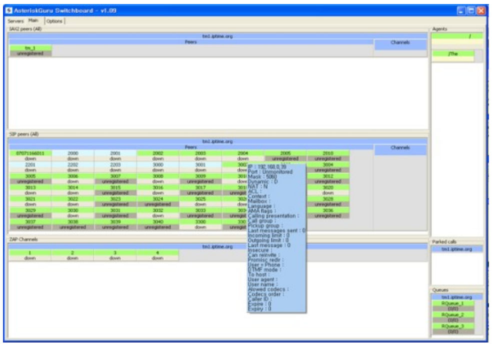
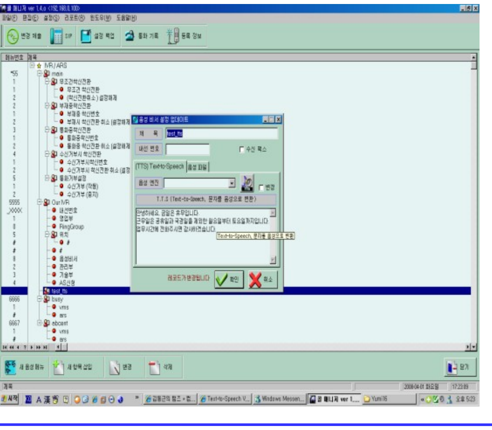
Supplied software (CRM, CDR Statistics)

Software for Call center agents and users

Category	Description
CRM	User friendly CTI enabled CRM software.  For Sales and Customer Support. (Language: Korean)  Standard CRM, Simplified version. (Language: Korean, English)
CDR Statistics	 Call Detail Report Statistics

IP-PBX Management, Monitoring and TTS software

Software for Administrators

Category	Description
IP-PBX Management software	Call Manager for IP-PBX Administration  GUI Call Manager runs for Windows OS. (Language: Korean)
Monitoring	 Operator Panel, Real-time Monitoring
TTS (TEXT TO SPEECH)	 Design ARS/IVR wit TTS editor. Requires SAPI 5.0 TTS engines.

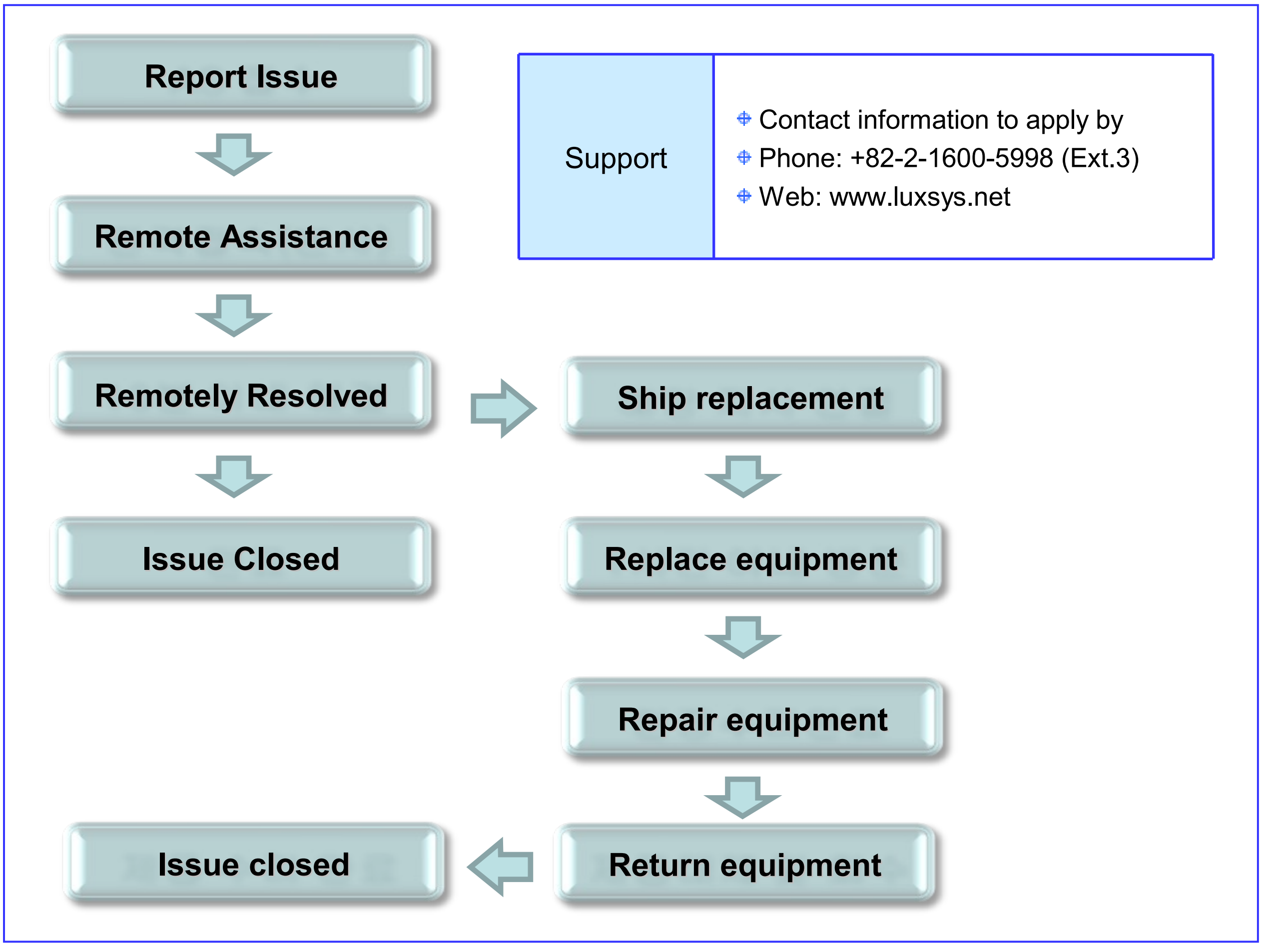
Maintenance and Support

We provide technical support to our customers.

The product comes with 1 year warranty, where customer can request replacement.

When issue cannot solved remotely, we ship the temporarily replacement while product is repaired.

Apply process for warranty support



Thank You !

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